AudioCodes Series of High Definition IP Phones

HD VoIP

Administrator's Manual

420HD IP Phones with Microsoft® Lync™ Server 2010

Version 2.0.1

AudioCodes

HD VoIP Sounds Better

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Notice

This manual shows administrators how to set up and configure AudioCodes' 420HD IP Phone in enterprise network environments.

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Related Documentation

Document Name
420HD IP Phone User's Manual
420HD IP Phone Quick Guide

Regulatory Information

Compliancy and Regulatory Information can be viewed at : http://www.audiocodes.com/library.

Note: Throughout this document, where features can be configured using the Web interface or configuration file, the Web parameter is displayed in the regular font above its corresponding configuration file parameter, which is enclosed in square brackets in bold font type.

1 Introduction

This Administrator's Manual is intended for system administrators responsible for setting up and configuring the AudioCodes 420HD IP Phone with Microsoft Lync, in enterprise network environments.

AudioCodes IP phones are based on AudioCodes' proprietary High Definition (HD) voice technology, providing clarity and a rich audio experience in Voice-over-IP (VoIP) calls.

The phones are fully-featured telephones that provide voice communication over an IP network, allowing you to place and receive phone calls, put calls on hold, transfer calls and so on.

The phones offer a wide variety of management and configuration tools:

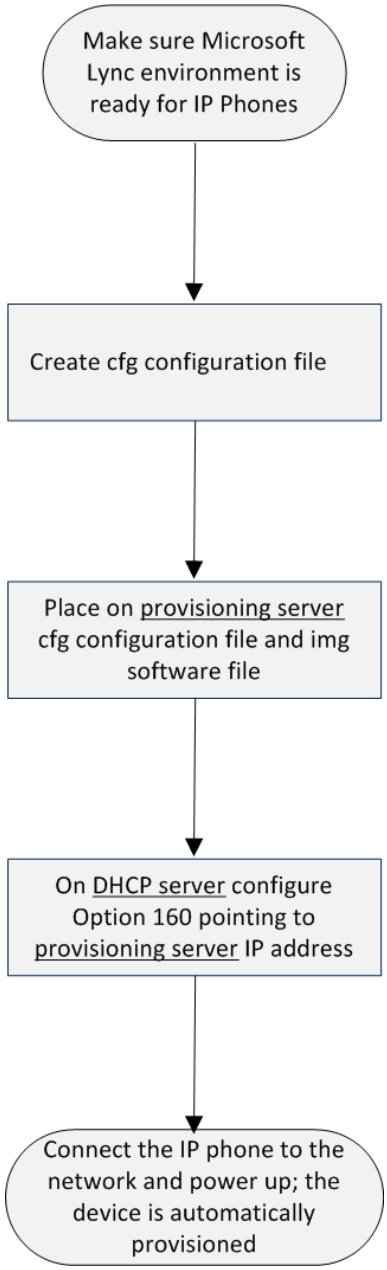
- **LCD**
  Easy-to-use, menu-driven display screen providing basic phone configuration and status capabilities
- **Web Interface**
  User-friendly user interface that runs on a Web browser (Microsoft Internet Explorer recommended).
- **Configuration File**
  Text-based file created using any plain text editor such as Microsoft's Notepad, containing configuration parameters. Loaded to the phone using the Web interface or a TFTP, FTP, HTTP or HTTPS server.
- **CLI over Telnet**

For detailed instructions on how to install the phone and operate its features, refer to the *User's Manual*.

1.1 Deployment Overview

The figure below shows how to deploy AudioCodes IP phones in an enterprise customer's network.

Figure 1-1: How to Deploy IP Phones in an Enterprise Customer's Network


Make sure Microsoft Lync environment is ready for IP Phones
Create cfg configuration file
Place on provisioning server cfg configuration file and img software file
On DHCP server configure Option 160 pointing to provisioning server IP address
Connect the IP phone to the network and power up; the device is automatically provisioned

2 Preparing the Microsoft Lync Environment for IP Phones

Before plugging in and playing the IP phones in an enterprise's Lync environment, make sure that the environment is ready for them.

To prepare it for IP phones, you must set up:

1. Front End Lync Server
2. Domain Controller, including:
   a. Active Directory, LDAP service
   b. DNS service
   c. DHCP service
   d. NTP service
3. Unified Messaging Server (optional)
4. Mediant™ Gateway
5. SBA Server (optional)

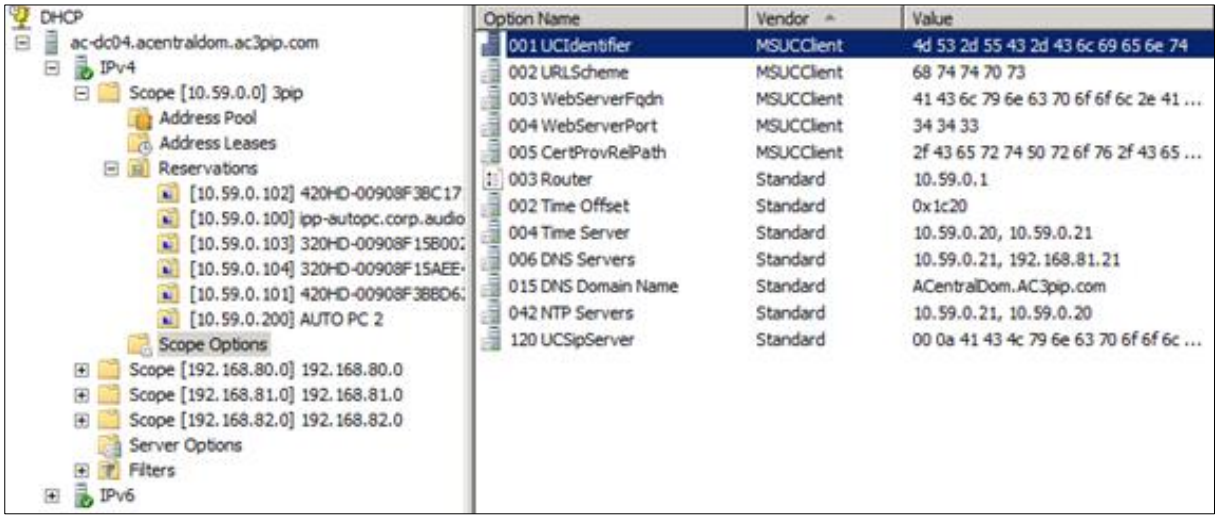
For details, refer to Microsoft's website at:
http://technet.microsoft.com/en-us/library/gg425854%28v=ocs.14%29.aspx

2.1 Making Sure DHCP Server Options are Correctly Configured

This section shows how to ensure that your enterprise's DHCP server options are correctly configured and that the network environment is ready for deployment of IP phones.

For detailed Microsoft instructions on setting up DHCP for the IP phone, see:
http://technet.microsoft.com/en-us/library/gg398369(v=ocs.14).aspx

Figure 2-1: DHCP Server Options


DHCP
ac-dc04.acentraldom.ac3pip.com
IPv4
Scope [10.59.0.0] 3pip
Address Pool
Address Leases
Reservations
[10.59.0.102] 420HD-00908F3BC17
[10.59.0.100] ipp-autopc.corp.audio
[10.59.0.103] 320HD-00908F15B00
[10.59.0.104] 320HD-00908F15AEE
[10.59.0.101] 420HD-00908F3BBD6
[10.59.0.200] AUTO PC 2
Scope Options
Scope [192.168.80.0] 192.168.80.0
Scope [192.168.81.0] 192.168.81.0
Scope [192.168.82.0] 192.168.82.0
Server Options
Filters
IPv6
Option Name
Vendor
Value
001 UCIdentifier MSUCClient 4d 53 2d 55 43 2d 43 6c 69 65 6e 74
002 URLScheme MSUCClient 68 74 74 70 73
003 WebServerFqdn MSUCClient 41 43 6c 79 6e 63 70 6f 6f 6c 2e 41 ...
004 WebServerPort MSUCClient 34 34 33
005 CertProvRelPath MSUCClient 2f 43 65 72 74 50 72 6f 76 2f 43 65 ...
003 Router Standard 10.59.0.1
002 Time Offset Standard 0x1c20
004 Time Server Standard 10.59.0.20, 10.59.0.21
006 DNS Servers Standard 10.59.0.21, 192.168.81.21
015 DNS Domain Name Standard ACentralDom.AC3pip.com
042 NTP Servers Standard 10.59.0.21, 10.59.0.20
120 UCSipServer Standard 00 0a 41 43 4c 79 6e 63 70 6f 6f 6c ...

Make sure that

- DHCP Option 43 (comprising 001-005 in the figure above) is correctly configured (see Section 2.1.1 on page 12 below)
- DHCP Option 120 is correctly configured (see Section 2.1.2 on page 12 below)
- DHCP Option 42 is correctly configured (see Section 2.1.3 on page 12 below)

Correct configuration of these three is critically important. The other DHCP options shown in the figure above are also important but are less susceptible to inaccuracies than these.

2.1.1 DHCP Option 43

Option 43 comprises the five sub-options 001-005 shown in the figure above and in the table below. These point the phone to the location of the Certificate Provisioning service on the Lync server. Use the table as a reference to make sure that each sub-option is correctly configured.

Refer also to http://technet.microsoft.com/en-us/library/gg398088(v=ocs.14).aspx

Table 2-1: DHCP Option 43 Configuration Reference

Sub-Option Number	Sub-Option Name	ASCII Value (example)
001	UCIdentifier	MS-UC-Client
002	URLScheme	https
003	WebServerFQDN	lyncserver.domain.com
004	WebServerPort	443
005	CertProvRelPath	/CertProv/CertProvisioningService.svc

2.1.2 DHCP Option 120

Option 120, which includes the Lync Server's fully qualified domain name (FQDN) as shown in the table below, is required for the certification authority (CA) pool Registrar.
Use the table as reference to make sure Option 120 is correctly configured.

Table 2-2: DHCP Option 120 Configuration Reference

Option Number	Option Name	ASCII Value (example)
120	UCSipServer	lyncserver.domain.com

2.1.3 DHCP Option 42

Option 42 specifies the servers that provide NTP /SNTP for the network. Make sure that NTP server IP addresses are correct, as shown in the table below.

Table 2-3: DHCP Option 120 Configuration Reference

Option Number	Option Name	String (example)
42	NTP Servers	10.59.0.20, 10.59.0.21

2.2 Making Sure the DHCP Server is Correctly Configured for Auto Provisioning

After creating a cfg configuration file (see Section 3 on page 17), place it - and the software file (img) and other files such as tone files - on a provisioning server from where the IP phones can download and install it.

To get the URLs to this provisioning server, the IP phones use DHCP. The provisioning server can be HTTP/S, TFTP or FTP.

The phone features *automatic update capability* to update the configuration and the software. Checks for newer configuration files and software versions are routinely automatically performed. Manual checks can also be performed.

➢ To make sure the feature functions correctly:

1. Verify that the provisioning server is running and that the configuration and firmware files are located in the correct location on it.
2. Connect the phone to the IP network and then to power.
3. On the DHCP server, configure DHCP Option 160 with the URL to the provisioning server where the configuration and firmware files are located.
   By default, the IP phone uses Option 160 which has highest priority.
   If absent, the IP phone uses Options 66/67 for TFTP.

   The following syntax is available for DHCP option 160:

   - <protocol>://<server IP address or host name>/<firmware file name>;<configuration file name>
   - <protocol>://<server IP address or host name>
   - <protocol>://<server IP address or host name>/<firmware file name>
   - <protocol>://<server IP address or host name>/;<configuration file name>

   Where <protocol> can be "ftp", "tftp", "http" or "https"
4. During DHCP negotiation, the phone requests DHCP options 66/67/160 to receive provisioning information. The DHCP server responds with Option 160 providing the provisioning URL, or Options 66 and 67 providing the TFTP IP address and firmware file name respectively.
5. The phone then checks whether new firmware is available by checking the firmware file header. If the version is different from the one currently running on it, the phone downloads the complete image and burns it to its flash memory.
6. If new firmware is unavailable, the phone checks whether a new configuration file is available on the server. If available, the phone downloads it and updates the phone's configuration after verifying that the configuration file is related to the phone model. When a configuration update is needed, the phone might reboot.

Notes:

- Only img (firmware) and cfg (configuration) files can be used.
- In the DHCP Discover message, the phone publishes its model name in Option fields 60 and 77 (e.g., 420HD). To provide different provisioning information to different models, set up a policy in the DHCP server according to phone model name.
- If the phone is powered off during provisioning, it becomes unusable; perform a recovery process (see Section 5 on page 69).
- Also see Section 4 on page 59 for more information on DHCP and on alternative provisioning methods.

2.3 Troubleshooting

2.3.1 Using the Phone LCD to Troubleshoot

Use the table below to help troubleshoot deployment problems that can occur after preparing the enterprise network environment for IP phone deployment.

Tip: Use the *first phone* that you deploy as an *indicator* for the entire deployment. If the first phone plugs in and plays without irregularities, all phones deployed after it should also. If it doesn't, troubleshoot as shown in this section before proceeding to deploy the other phones.

Note: After preparing the network and verifying readiness, make sure that the Lync PC client is operating, i.e., that the Lync server-client (Front End) setup is correct. Only after this, deploy the first phone.

Table 2-4: Troubleshooting Deployment Problems

Problem / LCD Notification	Corrective Action
Certificate problem LCD Notification: "Failed to validate certificate" -or- "Failed to obtain user certificate"	Three possible actions: ▪ Make sure that DHCP Option 43, sub-option 5, was enabled in the DHCP server. If it wasn't, enable it. ▪ Make sure you can access the Lync Web service URL: https://lyncsvrWebPoolFQDN:443/CertProv/CertProvisioningService.svc ▪ Query the LDAP server: _ldap._tcp.<DOMAIN name> Make sure it was enabled. If it wasn't, enable it in order to get the root certificate.
Synchronization problem. LCD Notification: "Failed to connect to time server" -or- "PIN internal error"	▪ Make sure that *at least one* of the following was configured to enable synchronization: ✓ NTP server, via DNS SRV record (_ntp._udp.<SIP domain>pointing to NTP server) ✓ NTP server, returned via DHCP Option 42 ✓ Time.windows.com ✓ Time.nist.gov ✓ Configuration Parameter (manually)
Phone not initializing	Make sure that DHCP is enabled.
Cannot find SIP server for 'Domain name'. Phone can't perform registration. LCD Notification: "Failed to connect <domain> server" -or- "Cannot find Lync server at <>"	▪ Make sure that *at least one* of the following is enabled in the DNS server: ▪ _sipinternaltls._tcp.<domain> (for TLS) ▪ DHCP results (Option 120) (for TLS) ▪ _sipinternal._tcp.<Domain> (for TCP) ▪ DHCP results (Option 120) (for TCP) ▪ _sip._tls.<Domain> (for TLS) ▪ _sip._tcp.<Domain> (for TCP)

Problem / LCD Notification	Corrective Action
LCD Notification: "Location look-up failed. Please enter your address."	▪ Make sure that 'Location look up' is configured by the management shell in the Lync server.
LCD Notification: "LAN Link failure"	The LAN link is disconnected. This is a general networking problem that's beyond the scope of this document. Either there's a physical cabling issue or there's a local or VLAN communications problem.
LCD Notification: "Duplicate IP"	This is a general networking problem. The IP address configured for this endpoint was already configured for another. ▪ In the DHCP server, delete the duplicate IP address and request another.
LCD Notification: "Failed to connect to Lync server"	This is a general networking problem beyond the scope of this document. If a communications problem occurs in the enterprise network, for example, if the server goes down, this notification is displayed on the phone LCD.
LCD Notification: "PIN invalid phone info"	The phone number or extension that was entered is invalid. ▪ Make sure that the correct information was entered in the phone LCD and in the Lync server interface, and that they tally. ▪ Verify in the Lync server interface that the PIN is enabled and if it isn't, enable it.
LCD Notification: "PIN not set"	▪ Verify in the Lync server interface that a PIN was configured for this user account. ▪ If it wasn't, create a PIN for the account.
LCD Notification: "PIN expired"	▪ In the Lync server interface, renew the PIN expiration policy.
LCD Notification: "PIN account disabled"	▪ Verify in the Lync server interface that the account was enabled. If it wasn't, enable it.
LCD Notification: "PIN internal error"	▪ Test the PIN Authentication process on the Lync server: Run in the server shell the emulate cmdlet: **Test-CsPhoneBootstrap -PhoneOrExt nnnn -PIN nnnn** ▪ If the test result is 'fail', there's a configuration error on the Lync server side, hence the PIN sign-in failure on the phone side. To troubleshoot, see: http://technet.microsoft.com/en-us/library/gg412852.aspx

Note:

- The *ringer* LED remains red until the problem is corrected.
- Users cannot dial or initiate calls if an LCD notification is displayed.

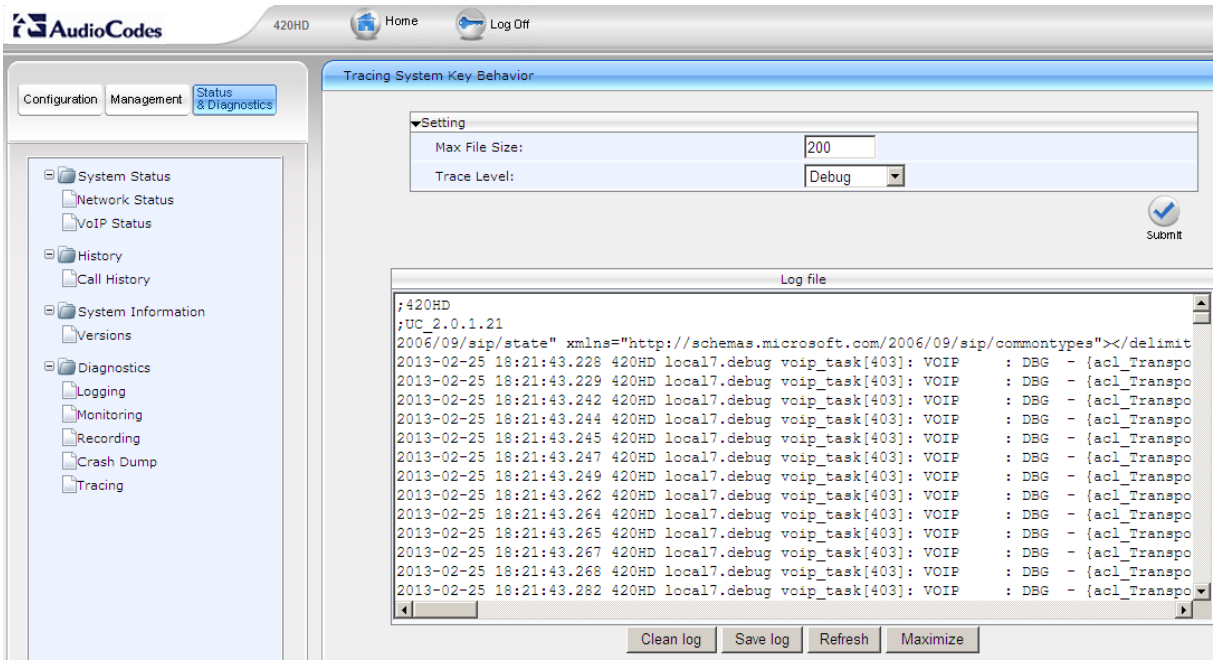
2.3.2 Using the Web Interface to Troubleshoot

This section shows how to use the Web interface to troubleshoot problems.

➢ To use the Web interface to troubleshoot:

1. Obtain the phone's IP address if the DHCP server successfully automatically provided one: Press the **MENU** key on the dial pad and in the phone's LCD, navigate to and select Status > Network Status
2. Scroll down to the IP Address and note it.
3. Open your browser and use the IP address to access the phone's Web interface.
4. Select the **Status & Diagnostics** tab > **Diagnostics** > **Tracing**:

Table 2-5: Web Interface – Status & Diagnostics - Tracing



3 Creating a Configuration File for Auto Provisioning via DHCP

Note: This section is intended for distributors / integrators responsible for configuring IP phones specific to customer countries / time zones / environments.

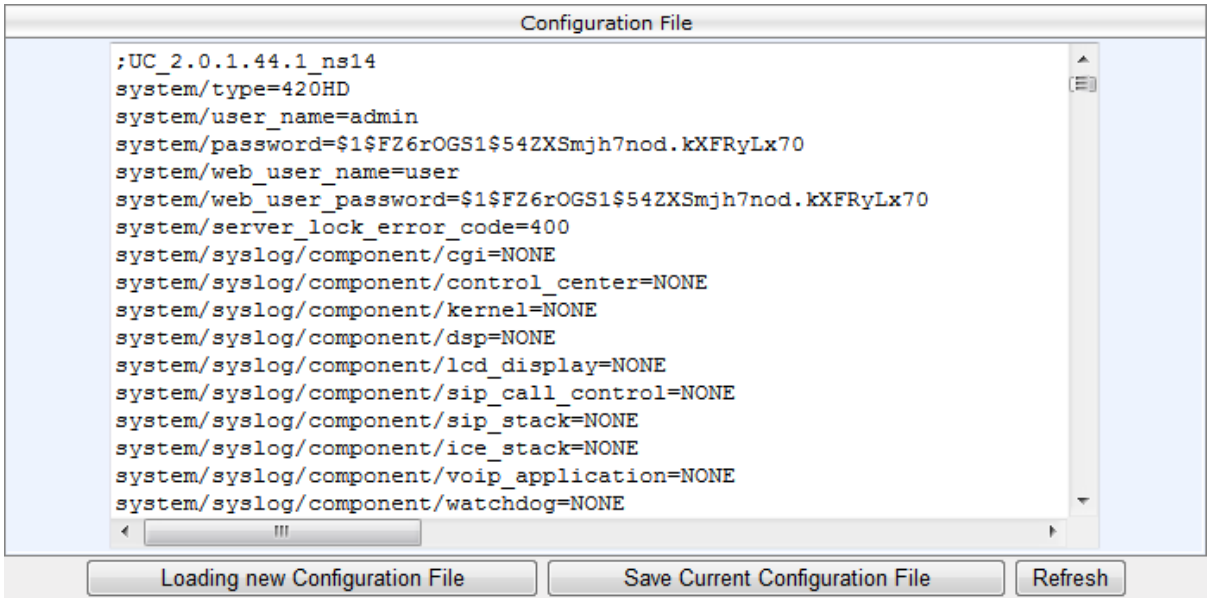
The sections below show how to configure a single IP phone.

After configuring, you'll then save the configuration as a cfg file on your PC and apply it to all IP phones deployed in the customer's enterprise, downloaded automatically from a provisioning server using DHCP (see Section 2.2 on page 13).

➢ To save the cfg file you configure:

1. Access the Configuration File page in the Web interface (**Management** tab > **Manual Update** menu > **Configuration File**):

Figure 3-1: Web Interface - Configuration File



2. Click the **Save Current Configuration File** button and save the cfg file in a folder on your PC.
3. Apply the cfg file to all IP phones in the customer's enterprise (see Section 2.2 on page 13).

Note: When creating a new configuration file, ensure that the *system/type* parameter in the cfg file is set to the proper phone model (e.g.,420HD).

Read the sections below for information on how to configure a single phone.

3.1 Configuring Network Settings

This section shows how to reconfigure IP network settings, including:

- LAN Connection Type (see Section 3.1.1 below)
- Partial DHCP (see Section 3.2 below)

3.1.1 Changing LAN Connection Type

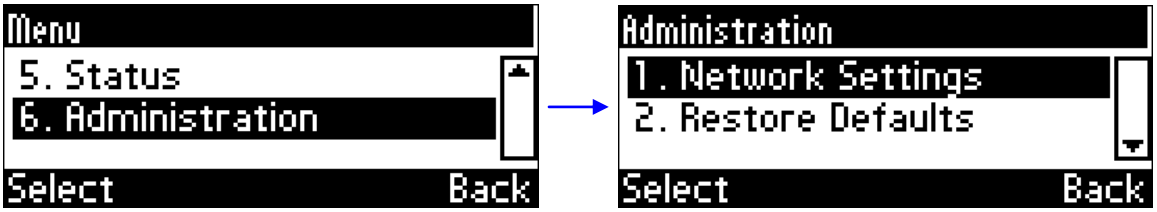
The phone's LAN Connection Type can be:

- Automatic IP (automatically provisioned by DHCP server from where the LAN IP address is obtained) (default)
- Static IP Address

This section shows how to change LAN Connection Type in the phone's LCD and through the Web interface.

➢ To change LAN Connection Type in the phone's LCD:

1. In the LCD access the Network Settings screen (**MENU** key > **Administration** > **Network Settings**):



2. Select **LAN Connection Type**:

3. Navigate to **Static IP**.
4. Define a static IP addressing scheme:
   a. Press the **Edit** softkey:

b. Press the **Edit** softkey:

c. Enter the new address in dotted-decimal notation, using the following keys:

- **Navigation control**: moves the cursor left or right in the IP address
- **Clear** softkey: deletes the digit to the left of the cursor.

d. Press the **Save** and then **Apply** softkey.

5. Navigate to and configure **Netmask**, **Gateway**, **Primary DNS** and **Secondary DNS** as you did **IP Address**.

➢ To change LAN Connection Type in the Web interface:

1. Access the Network Settings page (**Configuration** tab > **Network Connections** menu > **Network Settings**) and select the **Static IP** option:

Figure 3-2: Web Interface - Network Settings

2. Configure the parameters using the table below as reference and click **Submit**.

Table 3-1: Network Settings

Parameter	Description
Note: To add a value to these parameters, enter **network/** followed by the parameter name, equal sign and then the value (e.g. `network/lan_type`=DHCP).	
IP Type **[network/lan_type]**	Defines the IP addressing method: ▪ **[STATIC]** Static IP - Phone's IP address is defined manually ▪ **[DHCP]** Automatic IP DHCP (default) - Phone's IP address is acquired automatically from a DHCP server
network/lan/fixed_ip	This section defines parameters relevant if 'lan_type' is configured to **STATIC** or the corresponding 'network/lan/dhcp' parameter is set to **1**.
IP Address **[network/lan/fixed_ip/ip_address]**	The LAN IP address

Parameter	Description
Subnet Mask **[network/lan/fixed_ip/netmask]**	The subnet mask address
Default Gateway Address **[network/lan/fixed_ip/gateway]**	The IP address of the default gateway.
Domain Name **[network/lan/fixed_ip/domain_name]**	The domain name.
Domain Name Server (DNS)	
Primary DNS **[network/lan/fixed_ip/primary_dns]**	The primary DNS server address.
Secondary DNS **[network/lan/fixed_ip/secondary_dns]**	The secondary DNS server address. The phone connects to this server if the primary DNS server is unavailable.

3.2 Configuring DHCP Settings

In the Web interface's Network Settings page (see above), select the **Automatic IP (DHCP)** option. The following parameter settings can be configured in the Web interface and the configuration file:

Table 3-2: DHCP Parameters

Parameter	Description
Partial DHCP	
network/lan/dhcp	This sub-section defines the parameters to configure if **lan_type** is configured to "DHCP".
Domain Name - Manual **[network/lan/dhcp/domain_name/enabled]**	Enables setting the domain name manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/domain_name** must also be set.
IP Address - Manual **[network/lan/dhcp/ip_address/enabled]**	Enables setting the IP address manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/ip_address** must be set.
Subnet Mask - Manual **[network/lan/dhcp/netmask/enabled]**	Enables setting the network mask manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/netmask** must be set.
[network/lan/dhcp/gateway/enabled]	Enables setting the default gateway manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/gateway** must be set.
Primary DNS - Manual **[network/lan/dhcp/primary_dns/enabled]**	Enables setting the primary DNS manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/primary_dns** must be set.
Secondary DNS - Manual **network/lan/dhcp/secondary_dns/enabled**	Enables setting the secondary DNS manually. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** If enabled, **network/lan/fixed_ip/secondary_dns** must be set.

Parameter	Description
DHCP-Related Parameters	
network/lan/dhcp/ntp/server_list/enabled	Enables prioritization of the NTP server's information received from the DHCP server (Option fields 42 or 4), over the static configuration (**system/ntp/primary_server_address** and **system/ntp/secondary_server_address**). ▪ **[0]** Disable ▪ **[1]** Enable (default)
network/lan/dhcp/ntp/gmt_offset/enabled	Enables prioritization of the NTP GMT offset information received from the DHCP server (Option field 2), over the static configuration (**system/ntp/gmt_offset**). ▪ **[0]** Disable ▪ **[1]** Enable (default)

3.3 Configuring LAN and PC Port Settings

Port settings can be configured using the Web interface or the configuration file.

Note: The optional values of the configuration file parameters are enclosed in square brackets while its corresponding Web interface values are written outside the square brackets, for example, [1] Enable.

➢ To define the phone's port settings:

- **Using the Web interface:**

  1. Access the LAN Settings page (**Configuration** tab > **Network Connections** menu > **LAN Settings**).

Figure 3-3: Web Interface - Port Mode

  2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-3: Port Settings Parameters

Parameter	Description
LAN Port Mode **[network/lan/port_mode]**	Sets the LAN port mode. Valid values are : [AUTOMATIC] = Auto negotiation. [FULL_10] = 10Mbps + full duplex [FULL_100] = 100Mbps + half duplex [HALF_10] = 10Mbps + full duplex [HALF_100] = 100Mbps + half duplex
PC Port Mode **[network/pc/port_mode]**	Sets the computer port mode. See valid values above.

3.4 Configuring VLAN Settings

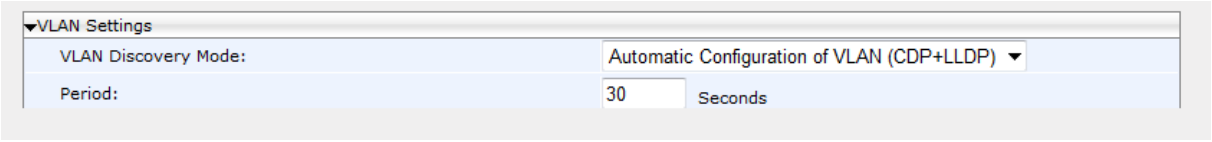
VLAN settings can be configured using the Web interface or the configuration file.

➢ **To configure the phone's VLAN settings:**

- **Using the Web interface:**

  1. Access the Network Settings page (**Configuration** tab > **Network Connections** menu > **Network Settings**).

Figure 3-4: Web Interface - VLAN Settings



  2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-4: VLAN Parameters Description

Parameter	Description
VLAN Discovery Mode **[network/lan/vlan/mode]**	Determines the VLAN mode of operation. ▪ **[Disable]** Disable ▪ **[Manual]** Manual Configuration of LAN - Static configuration of VLAN ID and priority ▪ **[CDP]** Automatic Configuration of VLAN - VLAN discovery mechanism based on Cisco Discovery Protocol (CDP) ▪ **[LLDP]** Automatic Configuration of VLAN - VLAN discovery mechanism based on LLDP. ▪ **[CDP_LLDP]** Automatic Configuration of VLAN (default) - VLAN discovery mechanism based on LLDP and Cisco Discovery Protocol (CDP). LLDP protocol is with higher priority.
Period **[network/lan/vlan/period]**	The time period in seconds between discovery messages when configured to CDP, LLDP or CDP and LLDP. The default value is 30.
VLAN ID **[network/lan/vlan/id]**	The VLAN ID. The valid range is 0 to 4096. The default VLAN ID is 0.
VLAN Priority **[network/lan/vlan/priority]**	The priority of traffic pertaining to this VLAN. The valid range is 0 to 7 (where 7 is the highest priority). The default VLAN priority is 0.

3.5 Enabling Telnet Access

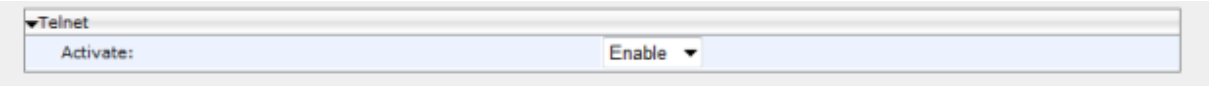
Telnet access can be enabled using the Web interface or the configuration file.

➢ **To enable Telnet:**

- **Using the Web interface:**

  1. Access the Telnet page (**Management** tab > **Remote Management** menu > **Telnet**).

Figure 3-5: Web Interface - Telnet



  2. Enable Telnet according to the parameter in the table below, and then click **Submit**.

- **Using the configuration file:**

Table 3-5: Telnet Parameters

Parameter	Description
Note: To add a value to these parameters, enter **management/** followed by the parameter name, equal sign and then the value (e.g. `management/telnet/enabled`=0).	
Activate **[management/telnet/enabled]**	Enables telnet access to the phone. ▪ **[0]** Disable (default) ▪ **[1]** Enable The user name and password for telnet access are according to the parameters: **system/user_name** and **system/password**.

3.6 Configuring VoIP Settings

This section shows how to configure VoIP settings.

3.6.1 Configuring Dialing Settings

Most dialing settings are automatically provisioned. This section shows how to configure dialing settings that aren't.

3.6.2 Configuring Tones

This section shows how to configure Tones using the Web interface or configuration file.

➢ **To configure tones:**

■ **Using the Web interface:**

1. Access the Dialing page (**Configuration** tab > **Voice Over IP** menu > **Dialing**).

Figure 3-6: Web Interface - Dialing Page

2. Access the Services page (**Configuration** tab > **Voice Over IP** menu > **Services**).

Figure 3-7: Web Interface - Services Page

3. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-6: Dial Tones Parameters

Parameter	Description
Dial Tone Timeout **[voip/dialing/dialtone_timeout]**	Defines the maximum duration of the dial tone (in seconds) after which the dial tone stops and a reorder tone is played. The valid range is 1 to 300. The default value is 30.
Reorder Tone Timeout **[voip/dialing/warning_tone_timeout]**	Defines the maximum duration of the reorder tone (in seconds) after which the reorder tone stops and a howler tone is played. The valid range is 1 to 300. The default value is 40.
Howler Tone Timeout **[voip/dialing/offhook_tone_timeout]**	Defines the duration (in seconds) of the howler tone. If the limit is exceeded, the howler tone stops. The howler tone indicates that the phone has been left in an off-hook state. The valid range is 1 to 300. The default value is 120.
Secondary Dial Tone **[voip/dialing/secondary_dial_tone/enabled]**	Enables the secondary dial tone. ▪ **[0]** Disable - Phone doesn't use secondary dial tone. ▪ **[1]** Enable (default) - Phone plays secondary dial tone if the secondary dial tone key is pressed (first digit).
Secondary Dial Tone Key **[voip/dialing/secondary_dial_tone/key_sequence]**	Defines the secondary dial tone is played if this is the first key pressed. The valid range is 0 to 9. The default value is 9. **Note:** This parameter is available only if the parameter **voip/dialing/secondary_dial_tone/enabled** is set to 1.
Out of Service Behavior **[voip/services/out_of_service_behavior]**	Determines whether a reorder tone is played instead of a dial tone if you configured a Registrar IP address and the registration failed. ▪ **[NONE]** No Tone ▪ **[REORDER_TONE]** Reorder Tone (default)
Stutter Tone Duration **[voip/services/msg_waiting/stutter_tone_duration]**	Defines the duration for which a stutter tone is played when you have unheard messages. The valid range is 1000 to 60000. The default value is 2500.
Automatic Disconnect **[voip/dialing/automatic_disconnect]**	Determines whether the phone automatically goes idle (i.e. on-hook) when the last remaining call is disconnected. This is only relevant when the speaker or headset is used. ▪ **[0]** Disable ▪ **[1]** Enable(default)

3.6.2.1 Loading Ring Tones

This section shows how to upload new ring tones using the Web interface or configuration file.

Notes:

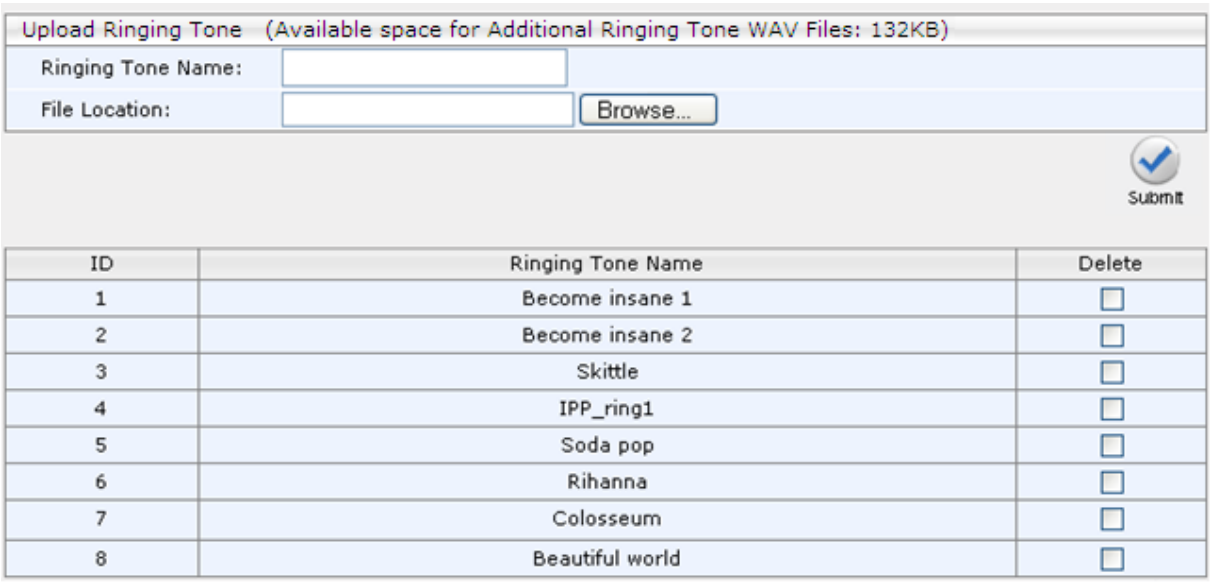
- The ring tone file must be in WAV file format (A/Mu-Law, 8-kHz audio sample rate and 8-bit audio sample size or PMC 16-kHz audio sample rate and 16-bit audio sample size, Intel PCM encoding).
- If you want the phone to use an uploaded ring tone, you need to use the phone's LCD screen to select it (refer to the phone's *User's Manual*).

➢ **To upload a ring tone:**

■ **Using the Web interface:**

1. Access the Tones page (**Configuration** tab > **Personal Settings** menu > **Tones**).

Figure 3-8: Web Interface - Uploading Ringing Tones



2. In the 'Ringing Tone Name' field, enter the name of the ring tone file that you want to upload. If you do not enter a name, the phone assigns the tone's file name (without the *.wav* file extension) as the name of the tone.
3. Click the **Browse** button, navigate to the folder in which the ring tone file is located, select the file, and then click **Open**; the file name and path is displayed in the 'File Location' field.
4. Click **Submit**; the file is loaded to the phone and displayed in the Ring Tone list.

➢ **To delete ring tones:**

- Deleting selected ring tones: select the 'Delete' check boxes corresponding to the ring tone that you want to delete, and then click **Submit**.

- **Using the configuration file:**

Table 3-7: Ring Tone Parameters

Parameter	Description
Ringing Tone Name File Location **[provisioning/ring_tone_uri]**	The URI for retrieving the ring tones file. The ring tones can be compressed to *.zip or *.tgz files and provided to the phone during provisioning. For example: **provisioning/ring_tone_uri=tones.tgz** **Notes:** ▪ The ringtone file is downloaded only after boot up, and not periodically. ▪ If the tones file is new, the phone updates the information, but does not reboot.

3.6.3 Configuring Regional Settings for Call Progress Tones

You can suite your phone's call progress tones to the location in which the phone is located. In addition, instead of using the default ring tone, you can upload a different ring tone file and use that ring tone for indicating incoming calls. This ring tone can only be selected using the phones LCD interface.

Follow the procedure below for selecting the region in which your phone is located. This is important for suiting your phone for the call progress tones of the country in which the phone is located.

➢ To select the geographical location of your phone:

- **Using the Web interface:**

1. Access the Tones page (**Configuration** tab > **Personal Settings** menu > **Tones**).

Figure 3-9: Web Interface - Tones - Regional Settings

2. From the 'Current Location' drop-down list, select the country in which your phone is located.
3. Click **Submit**.

- **Using the configuration file:**

Table 3-8: Regional Parameters

Parameter	Description
Current Location **[voip/regional_settings/selected_country]**	Defines the country in which your phone is located. The behavior and parameters of analog telephones lines vary between countries. The set of Call Progress Tones are all location-specific. The phone automatically selects the correct regional settings according to this setting. Supported countries: ▪ **[Israel]** Israel ▪ **[China]** China ▪ **[France]** France ▪ **[Germany]** Germany ▪ **[Netherlands]** Netherlands ▪ **[UK]** UK ▪ **[Brazil]** Brazil ▪ **[Italy]** Italy ▪ **[Argentina]** Argentina ▪ **[Portugal]** Portugal ▪ **[Russia]** Russia ▪ **[Australia]** Australia ▪ **[USA]** USA
voip/regional_settings/use_config_file_values	Enables the user-defined CPT. When this parameter is enabled, the **selected_country** parameter is not relevant and the below Call Progress Tones values can be determined by the user. ▪ **[0]** - Disable (default) ▪ **[1]** - Enable
Call Progress Tones (CPT) **Note:** Up to 10 CPT's can be configured (**voip/regional_settings/call_progress_tones/0…9**).	
voip/regional_settings/call_progress_tones/%d/enabled	Enables the specific CPT. ▪ **[0]** - Disable ▪ **[1]** - Enable
voip/regional_settings/call_progress_tones/%d/name	Defines the name of the CPT.
voip/regional_settings/call_progress_tones/%d/cadence	Defines the cadence type of the tone. ▪ **[0]** - Continuous signal ▪ **[1]** - Cadence signal ▪ **[2]** - Burst signal
voip/regional_settings/call_progress_tones/%d/frequency_a	Defines the low frequency (in Hz) of the tone. The valid value range is 300 to 1980 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.
voip/regional_settings/call_progress_tones/%d/frequency_b	Defines the high frequency (in Hz) of the tone. The valid value range is 300 to 3000 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.

Parameter	Description
voip/regional_settings/call_progress_tones/%d/frequency_a_level	Output level of the low frequency tone (in -dBm) in Call Progress generation. The valid range is 0 to 63, where 63 is mute.
voip/regional_settings/call_progress_tones/%d/frequency_b_level	Output level of the low frequency tone (in -dBm) in Call Progress generation. The valid range is 0 to 63, where 63 is mute.
voip/regional_settings/call_progress_tones/%d/tone_on_0	tone_on_0 to tone_on_3. If the signal is Cadence or Burst, then this value represents the on duration. If a Continuous tone, then this value represents the minimum detection time. The units are in 10 msec. The valid range is 0 to 10000.
voip/regional_settings/call_progress_tones/%d/tone_off_0	tone_off_0 to tone_on_3. If the signal is Cadence, then this value represents the off duration. The units are in 10 msec. If it is not used, then set it to zero. If the signal is Burst, only tone_off 0 is relevant. It represents the off time that is required from the end of the signal to the detection time. The valid range is 0 to 10000.

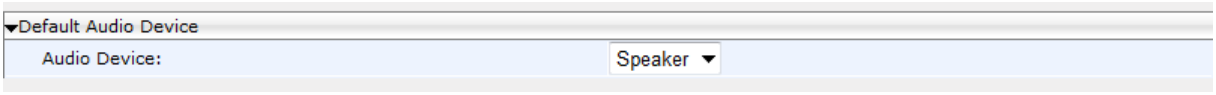
3.6.4 Configuring Default Audio Device

This section shows how to configure the default Audio Device using the Web interface or the configuration file.

➢ To configure the default Audio Device:

- **Using the Web interface:**
  1. Access the Dialing page (**Configuration** tab > **Voice Over IP** menu > **Dialing**).

Figure 3-10: Web Interface - Default Audio Device



  2. Configure the parameter according to the table below and then click **Submit**.
- **Using the configuration file:**

Table 3-9: Audio Device Parameter

Parameter	Description
Audio Device **[voip/answer_device]**	Sets the default audio device to answer or initiate a new call when no explicit audio device is set. For example: ▪ When pressing ANS SK. ▪ When initiating a call by speed dial key, call history or phone directory. ▪ Answering talk event or auto-answer. ▪ When starting to dial in “on hook” mode. Valid values are: ▪ [SPEAKER] ▪ [HEADSET]

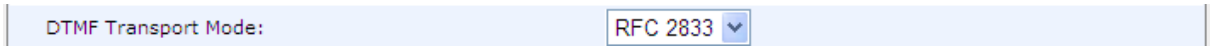
3.6.5 Configuring DTMF

This section shows how to configure DTMF using the Web interface or the configuration file.

➢ To configure DTMF:

- **Using the Web interface:**
  1. Access the Dialing page (**Configuration** tab > **Voice Over IP** menu > **Dialing**).

Figure 3-11: Web Interface - DTMF Transport Mode



  2. Configure using the table below as reference and click **Submit**.
- **Using the configuration file:**

Table 3-10: DTMF Description

Parameter	Description
DTMF Transport Mode **[voip/media/out_of_band_dtmf]**	DTMF transport mode. ▪ **[INBAND]** Inband ▪ **[RFC2833]** RFC 2833 (default) ▪ **[VIA_SIP]** Via SIP

3.7 Configuring Voice Settings

This section shows how to configure voice settings.

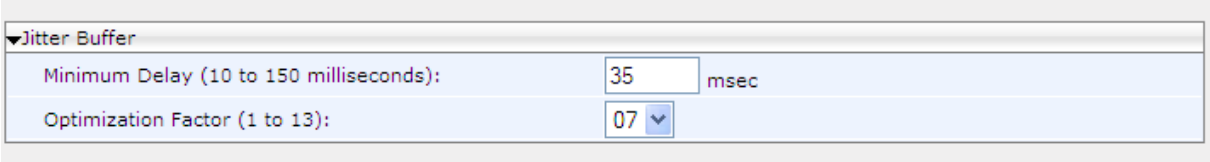
3.7.1 Configuring Jitter Buffer

This section shows how to configure Jitter Buffer using the Web interface or the configuration file.

➢ **To define jitter buffer parameters:**

1. Access the Voice page (**Configuration** tab > **Voice Over IP** menu > **Voice**).

Figure 3-12: Web Interface – Jitter Buffer



2. Configure the Jitter Buffer parameters according to the table below, and then click **Submit**.

■ **Using the configuration file:**

Table 3-11: Jitter Buffer Parameters

Parameter	Description
Minimum Delay **[voip/audio/jitter_buffer/min_delay]**	The initial and minimal delay of the adaptive jitter buffer mechanism, which compensates for network problems. The value should be set according to the expected average jitter in the network (in milliseconds). The valid range is 10 to 150. The default value is 35.
Optimization Factor **[voip/audio/jitter_buffer/optimization_factor]**	The adaptation rate of the jitter buffer mechanism. Higher values cause the jitter buffer to respond faster to increased network jitter. The valid range is 0 to 13. The default value is 7.

3.7.2 Configuring Silence Compression

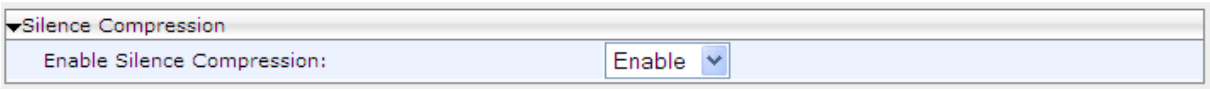
This section shows how to configure Silence Compression using the Web interface or the configuration file.

➢ To define silence compression:

- **Using the Web interface:**

1. Access the Voice page (**Configuration** tab > **Voice Over IP** menu > **Voice**).

Figure 3-13: Web Interface - Silence Compression



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-12: Silence Compression Parameters

Parameter	Description
Enable Silence Compression **[voip/audio/silence_compression/enabled]**	Enables silence compression for reducing network bandwidth consumption. ▪ **[0]** Disable (default) ▪ **[1]** Enable

3.7.3 Configuring Echo Cancellation

This section shows how to configure Echo Cancellation using the configuration file.

Table 3-13: Echo Cancellation Parameter

Parameter	Description
voip/audio/echo_cancellation/enabled	Enables echo cancellation. ▪ **[0]** - Disable ▪ **[1]** (default) – Enable **Note:** Disabling echo cancellation should be done for testing purposes only.

3.8 Activating Services

This section shows how to activate phone services.

3.8.1 Activating Automatic Dialing

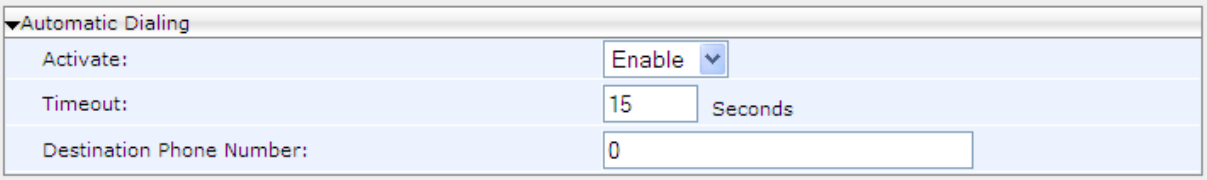
This section shows how to activate Automatic Dialing using the Web interface or the configuration file.

➢ **To activate automatic dialing:**

■ **Using the Web interface:**

1. Access the Dialing page (**Configuration** tab > **Voice Over IP** menu > **Dialing**).

Figure 3-14: Web Interface - Automatic Dialing



2. Configure using the table below as reference and click **Submit**.

■ **Using the configuration file:**

Table 3-14: Automatic Dialing Parameters

Parameter	Description
Activate **[voip/dialing/auto_dialing/enabled]**	Determines whether automatic dialing is enabled (i.e., phone number is automatically dialed when you off-hook the phone). ▪ **[0]** Disable (default) ▪ **[1]** Enable
Timeout **[voip/dialing/auto_dialing/timeout]**	Timeout (in seconds) before automatic dialing occurs after the phone is off-hooked. When set to 0, automatic dialing is performed immediately. The valid range is 0 to 120. The default value is 15.
Destination Phone Number **[voip/dialing/auto_dialing/destination]**	The number that is automatically dialed when the phone is off-hooked. The valid value can be up to 32 characters.

3.8.2 Enabling Automatic Answer

This section shows how to enable the Automatic Answer service using the configuration file parameters described in the table below.

Table 3-15: Configuration File - Automatic Answer Parameters

Parameter	Description

voip/auto_answer/enabled	Enables the Automatic Answering feature.
	▪ **[0]** Disable (default)
	▪ **[1]** Enable
	When this parameter is enabled and an incoming SIP INVITE message is received containing information that informs the IP phone to automatically answer the call, the phone answers the call immediately or after a timeout, depending on the auto-answer type specified in the INVITE message:
	▪ **Phone answers after a timeout:** The phone automatically answers the call after a timeout if the INVITE message includes a SIP Call-Info header with a tag value, "answer-after=" set to a number representing the timeout. During the timeout interval, the phone rings normally. If the call is answered or rejected during this interval, then the automatic answering mechanism is not used. However, if the phone is left to ring throughout the timeout interval, it automatically answers the call once this timeout expires.
	▪ **Phone answers immediately:** The phone answers the call immediately in any of the following cases:
	  ▪ If the SIP Alert-Info header contains the tag value, "ring answer".
	  ▪ If the SIP Alert-Info header contains the tag value, "info=alert-autoanswer".
	Notes:
	▪ If the SIP Call-Info header includes all the above answer types or any two different types (i.e., “answer-after=”, “ring answer”, and “alert-autoanswer”), the answer-after= type takes precedence.
	▪ If there is an existing call when an INVITE message for automatic answer is received, the existing call is automatically put on hold.

voip/talk_event/enabled	Enables the "talk" event feature. ▪ **[0]** Disable (default) ▪ **[1]** Enable The phone automatically answers an incoming call if it receives a SIP NOTIFY message with the "talk" event. If a call is already in progress, the call is put on hold and the incoming call is answered.

3.8.3 Activating Message Waiting Indication

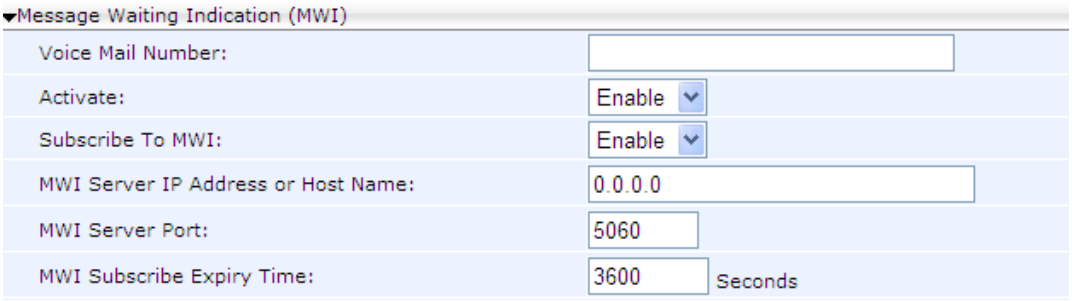
This section shows how to activate Message Waiting Indication (MWI) using the Web interface or the configuration file.

➢ **To activate MWI:**

- **Using the Web interface:**

1. Access the Services page (**Configuration** tab > **Voice Over IP** menu > **Services**).

Figure 3-15: Web Interface - MWI



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-16: MWI Parameters

Parameter	Description
Activate **[voip/services/msg_waiting_ind/enabled]**	Enables the MWI feature. ▪ **[0]** Disable ▪ **[1]** Enable (default)
Subscribe To MWI **[voip/services/msg_waiting_ind/subscribe]**	Determines whether the phone registers to an MWI server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
MWI Server Port **[voip/services/msg_waiting_ind/subscribe_port]**	The port number of the MWI server. Range = 1024-65535. Default = 5060.
MWI Server IP Address or Host Name **[voip/services/msg_waiting_ind/subscribe_address]**	The IP address or host name of the MWI server. Default = 0.0.0.0.
MWI Subscribe Expiry Time **[voip/services/msg_waiting_ind/expiration_timeout]**	Interval between MWI Subscribe messages. Range = 0-86400. Default = 3600.
Voice Mail Number **[voip/services/msg_waiting_ind/voice_mail_number]**	The extension number to access voice mail messages. Value = up to 64 characters.
voip/services/msg_waiting_ind/always_send_port	If the SIP port is the default port (i.e. 5060), then remove it from the Request-URI of the MWI SUBSCRIBE. ▪ **[0]** Disable ▪ **[1]** Enable (default)

3.9 Setting Volume Levels

This section shows how to set volume levels.

3.9.1 Configuring Automatic Gain Control

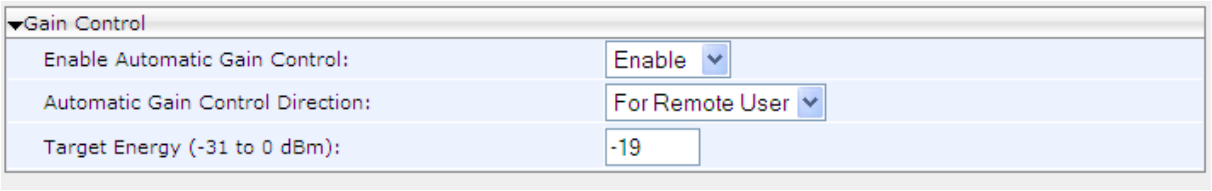
This section shows how to configure Automatic Gain Control using the Web interface or the configuration file.

➢ **To configure automatic gain control:**

- **Using the Web interface:**

1. Access the Voice page (**Configuration** tab > **Voice Over IP** menu > **Voice**).

Figure 3-16: Web Interface - Auto Gain Control



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-17: Automatic Gain Control Parameters

Parameter	Description
Enable Automatic Gain Control **[voip/audio/gain/automatic_gain_control/enabled]**	Enables the AGC. AGC automatically adjusts the phone's voice volume to compensate for weak or loud signals. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Automatic Gain Control Direction **[voip/audio/gain/automatic_gain_control/direction]**	Determines whether AGC is located before the encoder input (CTL_LOCAL) or after the decoder output (CTL_REMOTE). ▪ **[CTL_REMOTE]** For Remote User (default) - AGC is located after the Decoder output ▪ **[CTL_LOCAL]** For Local User **-** AGC is located before the Encoder input
Target Energy **[voip/audio/gain/automatic_gain_control/target_energy]**	The required output energy (in dBm) of the AGC. The valid range is -31 to 0 (dB). The default value is -19.

3.9.2 Configuring Tone Volume

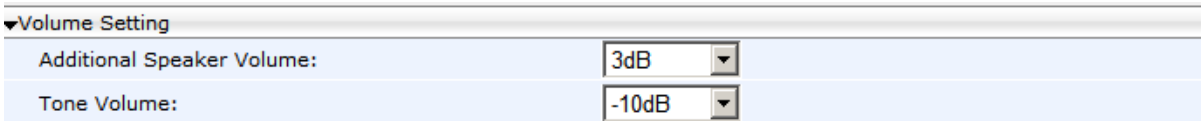
This section shows how to configure Tone volume using the Web interface or the configuration file.

➢ To configure tone volume:

- **Using the Web interface:**

1. Access the Volume Setting page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-17: Web Interface - Tone Volume



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-18: Tone Volume Parameter

Parameter	Description
Tone Volume **[voip/audio/gain/tone_signal_level]**	Call progress tone volume. This volume can be modified on-the-fly by pressing the phone's VOLUME key in certain scenarios. The valid range is 1 to 31 (-dB). The default value is 10 (-10dB).

3.9.3 Configuring Ringer Volume

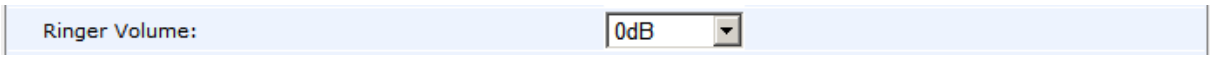
This section shows how to configure ringer volume using the Web interface or the configuration file.

➢ To configure the ringer volume:

- **Using the Web interface:**

1. Access the Volume Setting page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-18: Web Interface - Ringer Volume



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-19: Ringer Volume Parameters

Parameter	Description
Ringer Volume **voip/audio/gain/ringer_signal_level**	Ringing tone volume. This volume can be modified on-the-fly by pressing the phone's VOLUME key when the phone is in idle state. The valid range is [-31] to [31] dB

3.9.4 Configuring Speakerphone (Hands Free) Volume

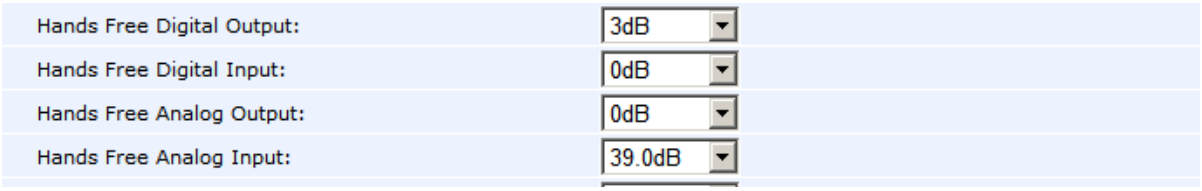
This section shows how to configure speakerphone (Hands Free) volume using the Web interface or the configuration file.

➢ **To configure speakerphone (hands free) volume:**

■ **Using the Web interface:**

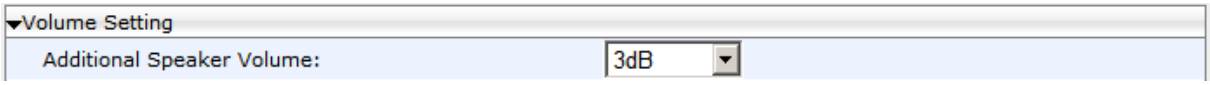
1. Access the Volume Setting page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-19: Web Interface - Hands Free



2. Configure using the table below as reference and click **Submit**.
3. Access the Services page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-20: Web Interface - Additional Speaker Volume



4. Configure using the table below as reference and click **Submit**.

Table 3-13: Speakerphone (Hands Free) Parameters

Parameter	Description
Hands-free Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word "minus" (e.g. =minus9db). ▪ Positive values: use the word "plus" (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
Hands Free Digital Output **[voip/audio/gain/handsfree_digital_output_gain]**	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Hands Free Digital Input **[voip/audio/gain/handsfree_digital_input_gain]**	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Hands Free Analog Output **[voip/audio/gain/handsfree_analog_output_gain]**	Analog output gain (in dB). Valid values: **[0db]**, **[minus1_5db]**, **[minus3db]**, **[minus4_5db]**, **[minus6db]**, **[minus7_5db]**, **[minus9db]**, **[minus10_5db]**, **[minus12db]**, **[minus13_5db]**, **[minus15db]**, **[minus16_5db]**, **[minus18db]**, **[minus19_5db]**, **[minus21db]**, **[minus22_5db]**, **[minus24db]**, **[minus25_5db]**, **[minus27db]**, **[minus28_5db]**, **[minus30db]**, **[minus31_5db]**, **[minus33db]**, **[minus34_5db]**, **[minus36db]**, **[minus37_5db]**, **[minus39db]**, **[minus39db]**, **[minus42db]**, **[minus48db]**, **[minus54db]**, **[MUTE]**
Hands Free Analog Input **[voip/audio/gain/handsfree_analog_input_gain]**	Analog input gain (in dB). Valid values: **[0db]**, **[plus1_5db]**, **[plus3db]**, **[plus4_5db]**, **[plus6db]**, **[plus7_5db]**, **[plus9db]**, **[plus10_5db]**, **[plus12db]**, **[plus13_5db]**, **[plus15db]**, **[plus16_5db]**, **[plus18db]**, **[plus19_5db]**, **[plus21db]**, **[plus22_5db]**, **[plus24db]**, **[plus25_5db]**, **[plus27db]**, **[plus28_5db]**, **[plus30db]**, **[plus31_5db]**, **[plus33db]**, **[plus34_5db]**, **[plus36db]**, **[plus37_5db]**, **[plus39db]**, **[plus40_5db]**, **[plus42db]**, **[plus48db]**, **[plus54db]**, **[MUTE]**
Additional Speaker Gain **[voip/audio/gain/additional_speaker_gain]**	Additional parameter for speaker gain configuration. ▪ **[0]** 0dB ▪ **[1]** 1dB ▪ **[2]** 2dB ▪ **[3]** 3dB

3.9.5 Configuring Handset Volume

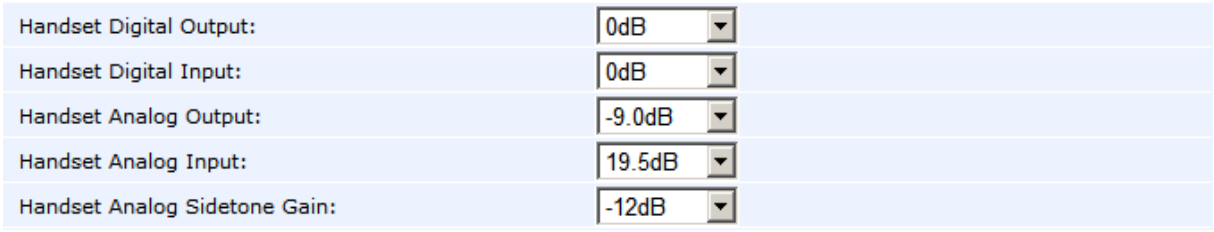
This section shows how to configure handset volume using the Web interface or the configuration file.

➢ To configure handset volume:

- **Using the Web interface:**

  1. Access the Volume Settings page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-21: Web Interface - Handset Volume Settings



  2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-14: Handset Gain Parameters

Parameter	Description
Handset Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word "minus" (e.g. =minus9db). ▪ Positive values: use the word "plus" (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
Handset Digital Output **[voip/audio/gain/handset_digital_output_gain]**	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Handset Digital Input **[voip/audio/gain/handset_digital_input_gain]**	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Handset Analog Output **[voip/audio/gain/handset_analog_output_gain]**	Analog output gain (in dB). Valid values: **[0db]**, **[minus1_5db]**, **[minus3db]**, **[minus4_5db]**, **[minus6db]**, **[minus7_5db]**, **[minus9db]**, **[minus10_5db]**, **[minus12db]**, **[minus13_5db]**, **[minus15db]**, **[minus16_5db]**, **[minus18db]**, **[minus19_5db]**, **[minus21db]**, **[minus22_5db]**, **[minus24db]**, **[minus25_5db]**, **[minus27db]**, **[minus28_5db]**, **[minus30db]**, **[minus31_5db]**, **[minus33db]**, **[minus34_5db]**, **[minus36db]**, **[minus37_5db]**, **[minus39db]**, **[minus39db]**, **[minus42db]**, **[minus48db]**, **[minus54db]**, **[MUTE]**

Parameter	Description
Handset Analog Input **[voip/audio/gain/handset_analog_input_gain]**	Analog input gain (in dB). Valid values: **[0db]**, **[plus1_5db]**, **[plus3db]**, **[plus4_5db]**, **[plus6db]**, **[plus7_5db]**, **[plus9db]**, **[plus10_5db]**, **[plus12db]**, **[plus13_5db]**, **[plus15db]**, **[plus16_5db]**, **[plus18db]**, **[plus19_5db]**, **[plus21db]**, **[plus22_5db]**, **[plus24db]**, **[plus25_5db]**, **[plus27db]**, **[plus28_5db]**, **[plus30db]**, **[plus31_5db]**, **[plus33db]**, **[plus34_5db]**, **[plus36db]**, **[plus37_5db]**, **[plus39db]**, **[plus40_5db]**, **[plus42db]**, **[plus48db]**, **[plus54db]**, **[MUTE]**
Handset Analog Sidetone Gain **[voip/audio/gain/handset_analog_sidetone_gain]**	Analog side tone gain (in db). Valid values: **[minus9db]**, **[minus12db]** (default), **[minus15db]**, **[minus18db]**, **[minus21db]**, **[minus24db]**, **[minus27db]**, **[MUTE]**

3.9.6 Configuring Headset Volume

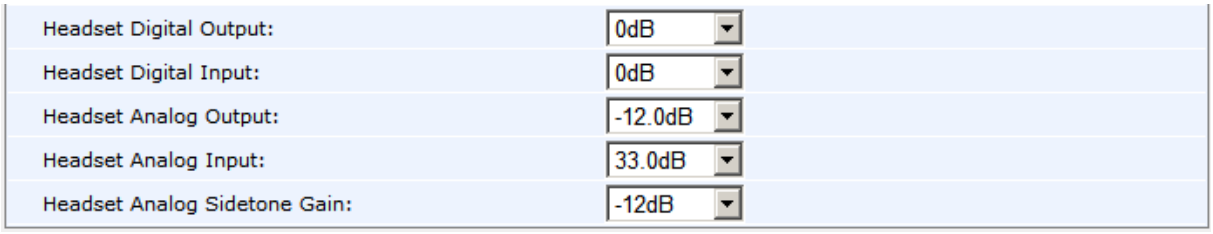
This section shows how to configure headset volume using the Web interface or the configuration file.

➢ To configure headset volume:

■ **Using the Web interface:**

1. Access the Volume Settings page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

Figure 3-22: Web Interface - Headset Fields



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-20: Headset Gain Parameters

Parameter	Description
Headset Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word "minus" (e.g. =minus9db). ▪ Positive values: use the word "plus" (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
Headset Digital Output **[voip/audio/gain/headset_digital_output_gain]**	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Headset Digital Input **[voip/audio/gain/headset_digital_input_gain]**	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute.
Headset Analog Output **[voip/audio/gain/headset_analog_output_gain]**	Analog output gain (in dB). Valid values: **[0db]**, **[minus1_5db]**, **[minus3db]**, **[minus4_5db]**, **[minus6db]**, **[minus7_5db]**, **[minus9db]**, **[minus10_5db]**, **[minus12db]**, **[minus13_5db]**, **[minus15db]**, **[minus16_5db]**, **[minus18db]**, **[minus19_5db]**, **[minus21db]**, **[minus22_5db]**, **[minus24db]**, **[minus25_5db]**, **[minus27db]**, **[minus28_5db]**, **[minus30db]**, **[minus31_5db]**, **[minus33db]**, **[minus34_5db]**, **[minus36db]**, **[minus37_5db]**, **[minus39db]**, **[minus39db]**, **[minus42db]**, **[minus48db]**, **[minus54db]**, **[MUTE]**
Headset Analog Input **[voip/audio/gain/headset_analog_input_gain]**	Analog input gain (in dB). Valid values: **[0db]**, **[plus1_5db]**, **[plus3db]**, **[plus4_5db]**, **[plus6db]**, **[plus7_5db]**, **[plus9db]**, **[plus10_5db]**, **[plus12db]**, **[plus13_5db]**, **[plus15db]**, **[plus16_5db]**, **[plus18db]**, **[plus19_5db]**, **[plus21db]**, **[plus22_5db]**, **[plus24db]**, **[plus25_5db]**, **[plus27db]**, **[plus28_5db]**, **[plus30db]**, **[plus31_5db]**, **[plus33db]**, **[plus34_5db]**, **[plus36db]**, **[plus37_5db]**, **[plus39db]**, **[plus40_5db]**, **[plus42db]**, **[plus48db]**, **[plus54db]**, **[MUTE]**
Headset Analog Sidetone Gain **[voip/audio/gain/headset_analog_sidetone_gain]**	Analog side tone gain (in db). Valid values: **[minus9db]**, **[minus12db]**, **[minus15db]**, **[minus18db]**, **[minus21db]**, **[minus24db]**, **[minus27db]**, **[MUTE]**

3.10 Configuring Advanced Phone Settings

3.10.1 Configuring the Phone Directory

This section shows how to configure the phone directory.

3.10.1.1 Configuring the Corporate Directory

This section shows how to configure the Corporate Directory using the Web interface or the configuration file.

3.10.1.1.1 Configuring an LDAP-based Corporate Directory

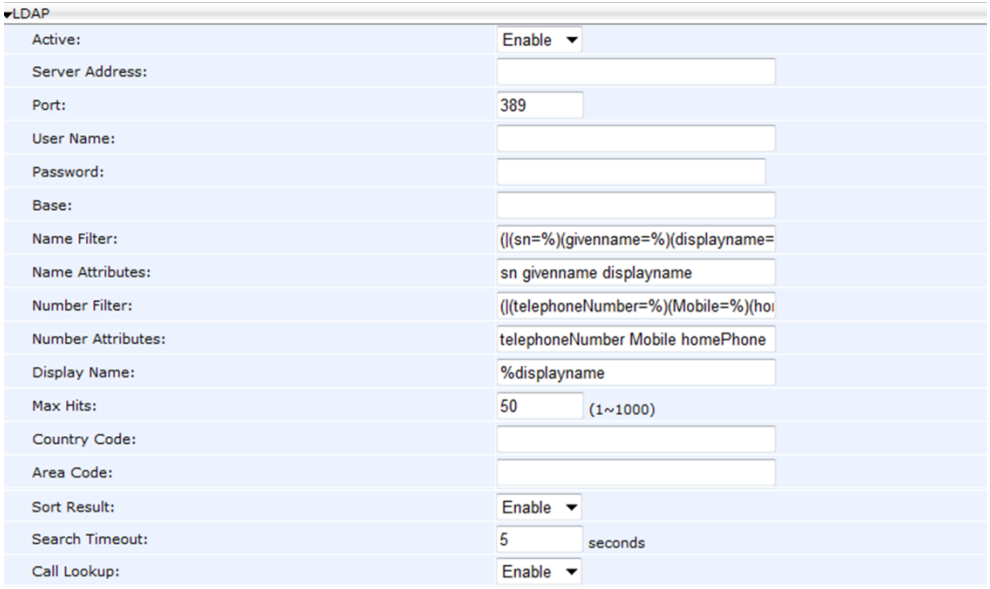
This section shows how to configure Lightweight Directory Access Protocol (LDAP) - an application protocol for accessing and maintaining distributed directory information services over an IP network. See RFC 4510 for a full description.

➢ **To configure LDAP:**

■ **Using the Web interface:**

1. Access the LDAP page (**Configuration** tab > **Advanced Applications** > **LDAP**). If LDAP is set to **Enable**, the page expands to show more parameters.

Figure 3-23: Web Interface - LDAP



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-21: LDAP Parameters

Parameter Name	Description
Active **[system/ldap/enabled]**	Enables or disable LDAP.
Server Address **[system/ldap/server_address]**	Defines the IP address or URL of the LDAP server.
Port **[system/ldap/port]**	Defines the LDAP service port.
User Name **[system/ldap/user_name]**	Defines the user name used for the LDAP search request.
Password **[system/ldap/password]**	Defines the password of the search requester.
Base **[system/ldap/base]**	Defines the access point on the LDAP tree.
Name Filter **[system/ldap/name_filter]**	Specifies your search pattern for name look ups. For example: When you type in the following field: *(&(telephoneNumber=*)(sn=%))*, the search result includes all LDAP records, which have the 'telephoneNumber' field set and the '("sn"-->surname)' field starting with the entered prefix. When you type in the following field: *(|(cn=%)(sn=%))*, the search result includes all LDAP records which have the ("cn"-->CommonName) OR ("sn"-->Surname) field starting with the entered prefix. When you type in the field *(!(cn=%))*, the search result includes all LDAP records which "do not" have the "cn" field starting with the entered prefix.
Name Attribute **[system/ldap/name_attrs]**	Specifies the LDAP name attributes setting, which can be used to specify the "name" attributes of each record which is returned in the LDAP search results. When you type in the following field, for example, *cn sn displayName*", this requires you to specify 'cn-->commonName'. This is the Full name of the user, sn-->Surname, last name or family name and "displayName" fields for each LDAP record.

Parameter Name	Description
Number Filter **[system/ldap/number_filter]**	Specifies your search pattern for number look ups. When you type in the following field, for example, *(|(telephoneNumber=%)(Mobile=%)(ipPhone=%))* , the search result is all LDAP records which have the “telephoneNumber” OR “Mobile” OR “ipPhone”field match the number being searched. When you type in the following field: *(&(telephoneNumber=%)(sn=*))*, the search result is all LDAP records which have the “sn” field set and the “telephoneNumber” match the number being searched.
Number Attribute **[system/ldap/number_attrs]**	Specifies the LDAP number attributes setting, which can be used to specify the “number” attributes of each record which is returned in the LDAP search results. When you type in the following field, for example, *Mobile telephoneNumber ipPhone*, you must specify ‘Mobile’, ‘telephoneNumber’ and ‘ipPhone’ fields for each LDAP record.
Display Name **[system/ldap/display_name]**	Specifies the format in which the “name, e.g. “Mike Black” of each returned search result is displayed on the IPPHONE. When you type in the following field, for example:**%sn, %givenName**, the displayed result returned should be “Black, Mike”.
Max Hits **[system/ldap/max_hits]**	Specifies the maximum number of entries expected to be sent by the LDAP server (this parameter is sent to the LDAP server).
Sort Result **[system/ldap/sorting_result]**	Sorts the search result by display name on the client side.
system/ldap/predict_text	This parameter appears in the configuration file; however, it is currently not supported.
Search Timeout **[system/ldap/search_timeout]**	The time out value for LDAP search (this parameter is sent to the LDAP server).
system/ldap/ui/use_right_arrow_active_search	This parameter appears in the configuration file; however, it is currently not supported.
system/ldap/lookup_incoming_call	This parameter appears in the configuration file; however, it is currently not supported.
Call Lookup **[system/ldap/call_lookup]**	Performs an LDAP search during call (search the display name for a number).
Country Code **[system/ldap/country_code]**	Defines the country code prefix added for number search.
Area Code **[system/ldap/area_code]**	Defines the area code prefix added for number search.

Parameter Name	Description
system/ldap/minimal_name_search_length	Starts to perform an LDAP search after x characters are input.
system/ldap/send_queries_while_typing	Sends an LDAP search each time the user presses a key (all keys with both number and letters).

3.10.1.1.2 Loading a Text-based Corporate Directory File

The Configuration file can include a link to a user-defined Corporate Directory file, using the **provisioning/corporate_directory_uri** parameter. This allows you to upload a corporate directory to the phone.

The Corporate Directory file includes a list of contacts and their phone numbers. The file must be a simple text file that can be created using an Excel document and saved as a CSV file.

The syntax of the corporate directory file must be as follows:

```
<full name>,<office>,<home>,<mobile>
```

For example:

```
John Smith,1234,98765432,574685746
```

If not all phone numbers are required, the relevant field must be left empty. For example, in the below directory entry, the home and user-defined numbers are absent:

```
John Smith,1234,,574685746
```

➢ **To configure the corporate directory:**

- **Using the Web interface:**

1. Prepare the TXT-based file as explained above.
2. Access the Directory page (**Configuration** tab > **Personal Settings** menu > **Directory**).
3. Click **Browse** and select the file to upload.
4. Click **Load Corporate Directory**.

Figure 3-24: Web Interface - Corporate Directory

5. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-22: Provisioning Parameters

Parameter	Description
Note: To add a value to these parameters, enter **`provisioning/`** followed by the parameter name equals the value (e.g. `provisioning/method=dynamic`).	
provisioning/corporate_directory_uri	The URI for retrieving the corporate directory. The corporate directory must be included in a separate file that can be downloaded to the phone during provisioning. For example: **provisioning/corporate_directory_uri=corporate_dir.txt** **Notes:** ▪ The corporate directory file is downloaded after boot up and periodically. ▪ If the corporate directory file is new, the phone updates the information, does not reboot.

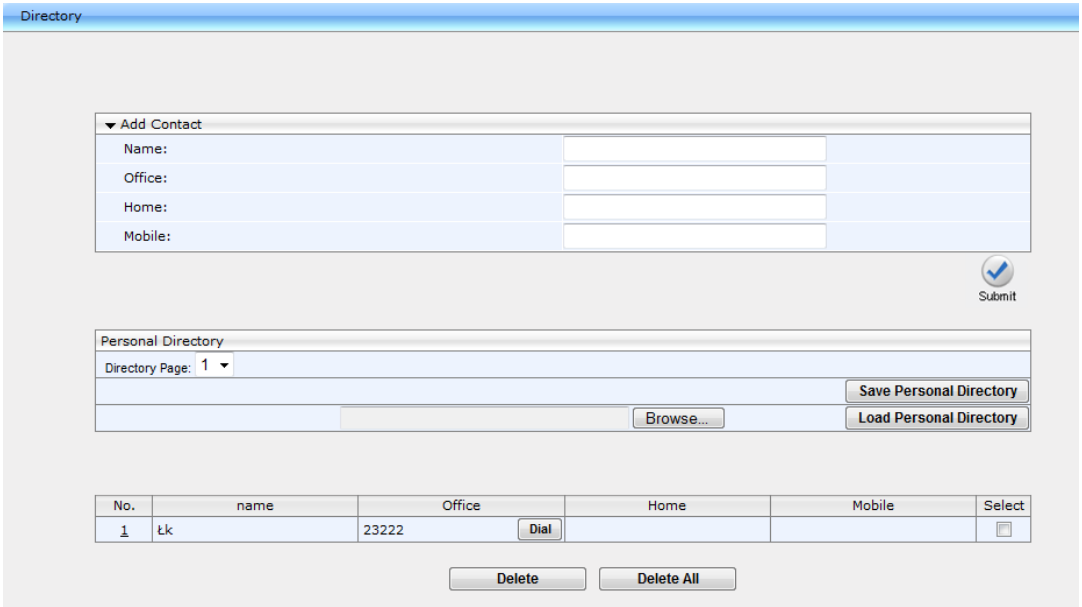
3.10.2 Managing the Local Phone Directory

You can add, edit, or delete directory contacts. A contact's address can be a telephone number, IP address, or domain name. You can also download or upload a personal directory file via the Web interface.

➢ To add a contact to the phone's directory:

1. Access the Directory page (**Configuration** tab > **Personal Settings** menu > **Directory**).

Figure 3-25: Web Interface - Directory - Add Contact



2. Under the **Add Contact** group, define the contact:
3. In the 'Name' field, enter the name of the contact.

4. In the 'Office', 'Home' and/or 'Mobile' fields, enter the contact's telephone numbers. The contact's number can be defined with an IP address or domain name (e.g. <number>@<IP address or domain name>).
5. Click **Submit**; the contact appears in the Directory list at the bottom of the page.

➢ **To edit a contact:**

1. If the contact does not appear in the displayed Directory list, then from the 'Directory Page' drop-down list, select the page in the directory that you want displayed.
2. In the Directory list, click the number that appears in the 'No.' column corresponding to the contact you want to edit; the contact's attributes appear in the **Edit Phone** group above.
3. Edit the contact as required, and then click **Submit**; the contact's new attributes are updated in the Directory list.

➢ **To delete a contact:**

1. In the Directory list, mark the 'Select' check box corresponding to the contact you want to delete.
2. Click **Delete**. (To delete all contacts, click the **Delete All** button.)

3.11 Configuring Speed Dial

This section shows how to configure Speed Dial using the Web interface or the configuration file.

Note: The phone's speed dials can also be defined in a simple text-based editor, placed on a server (e.g., HTTP or FTP/TFTP), and then uploaded to the phone using a configuration file. For a detailed description, see Section 3.11.3.

3.11.1 Adding a Speed Dial

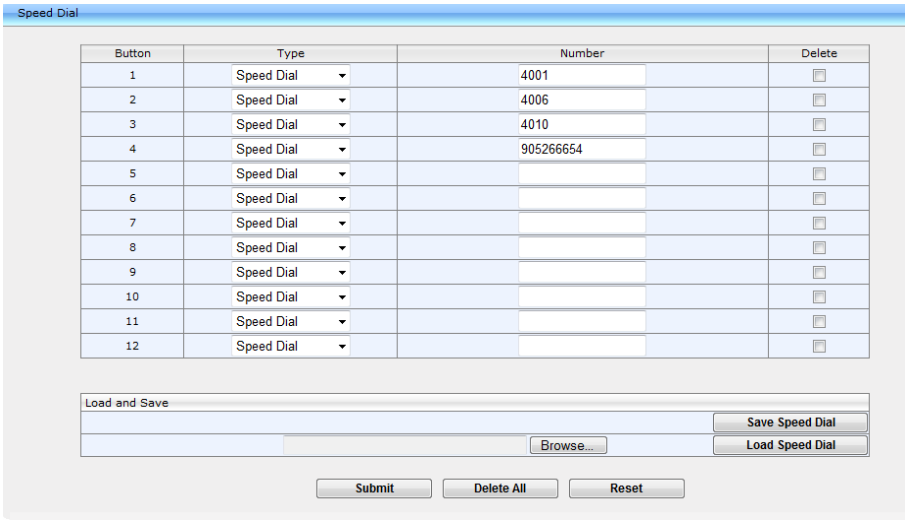
Adding a Speed Dial can be configured using the Web interface or the configuration file.

➢ **To add a speed dial:**

■ **Using the Web interface:**

1. Access the Speed Dial page (**Configuration** tab > **Personal Settings** menu > **Speed Dial**).

Figure 3-26: Web Interface - Speed Dial



2. In the 'Number' field corresponding to the phone's Speed Dial key (in the 'Button' column), enter the speed dial number to which you want to assign the Speed Dial key.
3. Click **Submit**.

■ **Using the configuration file:**

Table 3-23: Speed Dial Parameters

Parameter Name	Description
provisioning/speed_dial_uri	The URI for retrieving the speed dial list. The speed dial list must be included in a separate file that can be downloaded to the phone during provisioning. For example: **provisioning/speed_dial_uri=speed_dial_list.txt** **Notes:** ▪ The speed dial file is downloaded after boot up and periodically. ▪ If the speed dial file is new, the phone reboots. ▪ For creating a Speed Dial file, see Section 3.11.3.

3.11.2 Deleting a Speed Dial

➢ **To delete speed dials, either:**

- **Deleting selected speed dial entries:** select the 'Delete' check boxes corresponding to the speed dials that you want to delete and click **Submit**.
- **All speed dials:** Click **Delete All** and at the prompt click **OK**.
- **To clear (unselect) all selected 'Delete' check boxes:** click **Reset**.

3.11.3 Saving Speed Dial File Configuration to a File

Once you have configured your Speed Dial in the Web interface, you can save it to a file on your computer. This may be useful for backing up your configuration or for using the same configuration by loading it to another phone.

➢ **To save a speed dial:**

1. Access the Speed Dial page (**Configuration** tab > **Personal Settings** menu > **Speed Dial**).
2. Click **Save Speed Dial**.

➢ **To load a speed dial:**

1. Access the Speed Dial page (**Configuration** tab > **Personal Settings** menu > **Speed Dial**).
2. Click **Browse….** to select the *.cfg* file.
3. Click **Open**; the selected *.cfg* filename and path appear on the Web interface alongside the **Browse…** button.
4. Click **Load Speed Dial**.

3.11.4 Creating a Speed Dial File for Configuration File

The Configuration file can include a link to a user-defined Speed Dial file, using the **provisioning/speed_dial_uri** parameter. This allows you to upload speed dial settings to the phone.

The Speed Dial file must include a list of speed dial configurations. The file must be a simple text file that can be created using an Excel document and saved as a CSV file.

The syntax of the speed dial file is as follows:

```
<memory key>,<speed dial phone number>,<type>
```

where:

- *memory key* denotes the speed dial memory key on the phone.
- *speed dial phone number* denotes the phone number that is automatically dialed, when the user presses the speed dial key.
- *type* denotes the Speed Dial feature and must be set to “0”.

Below is an example of a Speed Dial file:

```
1,4418,0
2,4403,0
3,039764432,0
4,4391,0
12,1234,0
```

3.12 Configuring Security

3.12.1 Using the Encryption Tool

This section shows how to use the encryption tool.

3.12.1.1 Encrypting Configuration Files

This section shows how to encrypt the configuration file when, for example, it is sent over an unsecure network.

➢ **To encrypt the configuration file:**

- At the command line prompt, specify the following:

```
encryption_tool.exe -f <filename>.cfg
```

where <*file name*>.cfg specifies the name of the Configuration file that you wish to encrypt.

Once the Configuration file is encrypted, it receives the suffix '.cfx' (e.g. Conf.cfx). This is the file that you should specify in the 'Configuration URL' and the 'Dynamic Configuration URL' fields when performing automatic provisioning (see Part II 'Automatic Provisioning').

3.12.2 Encrypting Passwords in Configuration File

This section shows how to encrypt IP phone passwords used in the configuration process, for example, the 'System' password and the 'SIP Authentication' password.

➢ **To encrypt passwords:**

1. At the command line prompt, specify the following:

```
encryption_tool.exe -s <password_string>
```

where <*password_string*> specifies the string of the password that you wish to encrypt.

Once the password is encrypted, a string is generated with the following syntax:

```
{"<encrypted_string>"}
```

For example:

```
{"0qrNRpSJ6aE="}
```

2. Copy the generated string (including the {" "}) with the syntax specified above to the relevant parameter in the Configuration file.

For example, if you encrypted the SIP authentication password, the following is displayed in the relevant line in the configuration file:

```
voip/line/0/auth_password={"0qrNRpSJ6aE="}
```

Note: It is recommended to encrypt the System password using this procedure. If you choose not to do so, then the System password is by default encrypted using MD5.

3.12.3 Configuring 802.1X

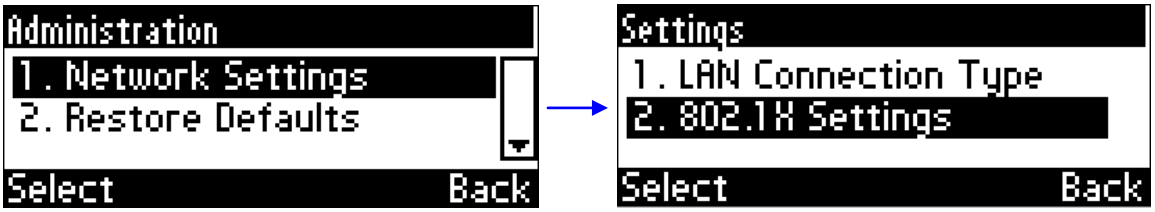
802.1X is an IEEE Standard for port-based Network Access Control (PNAC). It's part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism for devices joining a LAN or WLAN.

3.12.3.1 Using the Phone LCD

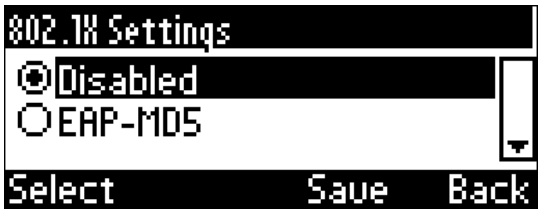
This section shows how to configure 802.1X from the phone LCD.

➢ **To configure 802.1X:**

1. Access the 802.1X Settings screen (MENU key > **Administration** > **Network Settings**).



2. Navigate to and select **802.1X Settings**:



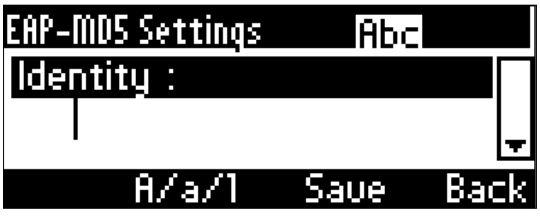
3. Navigate to and select either:
   - Disabled – disables the 802.1X feature
   - EAP-MD5 – see Section 3.12.3.1.1
   - EAP-TLS - see Section 3.12.3.1.2

3.12.3.1.1 EAP-MD5 Mode

This section shows how to configure EAP-MD5 mode for 802.1X.

➢ **To configure EAP-MD5 mode for 802.1X:**

1. Navigate to the **EAP-MD5** option and press **Select** and then **Edit**:



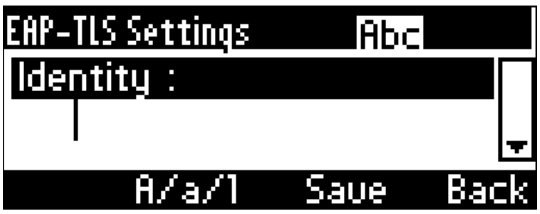
2. Enter this information:
   - **Identity:** User ID
   - **Password:** MD5 password (optional)
3. Press the **Save** softkey; a message appears notifying you that the phone will restart.
4. Press **Apply**.

3.12.3.1.2 EAP-TLS Mode

This section shows how to configure EAP-TLS mode for 802.1X.

➢ **To configure EAP-TLS mode for 802.1X:**

1. Navigate to the **EAP-TLS** option and press **Select** and then **Edit**:



2. Enter the following information:
   - **Identity:** user ID.
   - **Root Certificate URL:** full URL of the file which contains root certification file.
   - **Trusted Certificate URL:** full URL of the file which contains trusted certification file.
   - **Local Certificate URL:** full URL of the file which contains local certification file.
   - **Private Key URL:** full URL of the file which contains the private key file.
   - **Private key Password:** (option) In case the private key file is encrypted you should input the password to decrypt the file)
3. Press the **Save** softkey; a message appears notifying you that the phone will restart.
4. Press **Apply**.

3.12.3.2 Using the Web Interface / Configuration File

This section shows how to configure 802.1X from the Web interface or using the configuration file.

3.12.3.2.1 EAP MD5 Mode

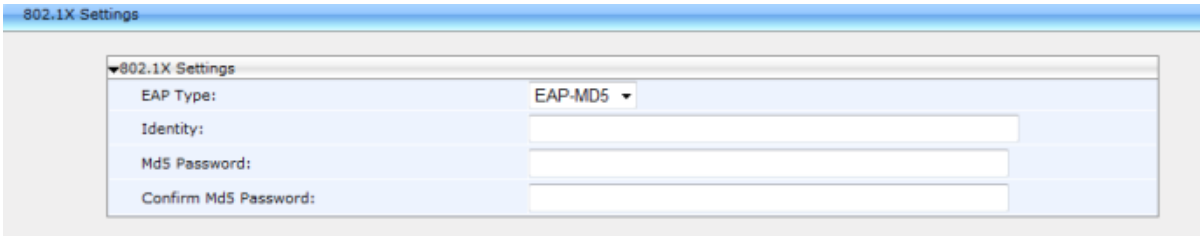
This section shows how to configure 802.1X settings for EAP-MD5.

➢ **To configure the phone's 802.1X settings for EAP-MD5:**

■ **Using the Web interface:**

1. Access the 802.1X Settings page (**Network Connections** > **802.1X Settings**).

Figure 3-27: Web Interface - 802.1X Settings - EAP MD5



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-24: EAP MD5 Parameters

Parameter	Description
EAP Type **[/network/lan/_802_1x/eap_type]**	Sets 802.1X Extensible Authentication Protocol mode [Disable] = Disables the use of 802.1X [EAP_MD5]=Authentication is implemented by user name and password (Password is optional). [EAP_TLS]= Authentication is implemented by Certificate, Client Certificate and Client Private Key.
Identity **[/network/lan/_802_1x/md5_identity]**	User ID for md5 mode.
MD5 password **[/network/lan/_802_1x/md5_password]**	Password for md5 mode. (Leave blank if no password).

3.12.3.2.2 EAP TLS Mode

This section shows how to configure phone's 802.1X settings for EAP-TLS.

➢ To configure the phone's 802.1X settings for EAP-TLS:

- **Using the Web interface:**

1. Access the 802.1X Settings page (**Network Settings** > **802.1X Settings**).

Figure 3-28: Web Interface - 802.1X Settings - EAP TLS

2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 3-25: EAP TLS Parameters

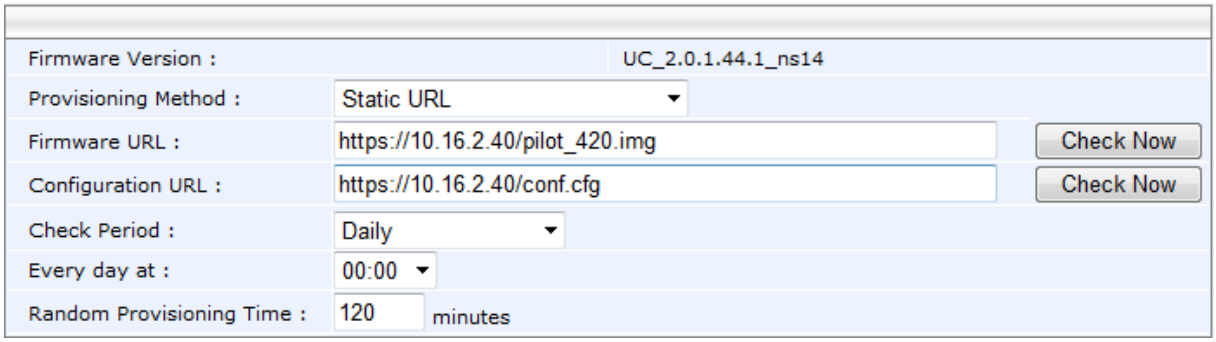
Parameter	Description
Identity **/network/lan/_802_1x/tls_identity**	Identity for TLS mode. (optional)
Root and Intermediate Certificate Filename **[/network/lan/_802_1x/root_ca_url]**	URL for the file that contains root and intermediate Certificate.
Trusted Certificate Filename **[/network/lan/_802_1x/trust_ca_url]**	URL for the file that contains a trusted Certificate.
Local Certificate Filename **[/network/lan/_802_1x/local_ca_url]**	URL for the file that contains a local Certificate.
Private Key Filename **[/network/lan/_802_1x/private_key_url**	URL for file that contains the private key.
Private Key Password **[/network/lan/_802_1x/private_key_password]**	Optional: If the file containing the private key is encrypted, this is the password for decryption.

3.12.4 Configuring HTTPS

This section shows how to access the Web interface using HTTPS, for example, https://10.16.2.40, to secure provisioning.

Firmware and configuration files will then be loaded to the phone from the HTTPS server. The Firmware URL and Configuration URL parameters in the screen below show how HTTPS is defined:

Figure 30-1: Web Interface – HTTPS



Note: To implement secure provisioning using HTTPS, the HTTPS server on the far end (from where you are loading the files) must also support HTTPS.

4 Configuring Automatic Provisioning

AudioCodes' IP phone is by default shipped ready for out-of-the-box deployment with *automatic provisioning capabilities*.

The phone also features an integral *automatic update mechanism* to automatically remotely upgrade its software image and update its configuration. Checks for newer software versions and configuration files are routinely automatically performed. Manual checks can also be performed.

You can load various files to the IP phone, such as CFG files, Certification Files, Speed Dial files, and Tones files, using protocols:

- HTTP/S
- TFTP
- FTP

AudioCodes' IP phone is by default shipped ready for out-of-the-box deployment with *automatic provisioning capabilities*.

Administrators can use the Web interface to configure *automatic* provisioning by:

- DHCP
- Static DNS Record
- SIP SUBSCRIBE and NOTIFY Messages

4.1 Configuring Automatic Provisioning via DHCP

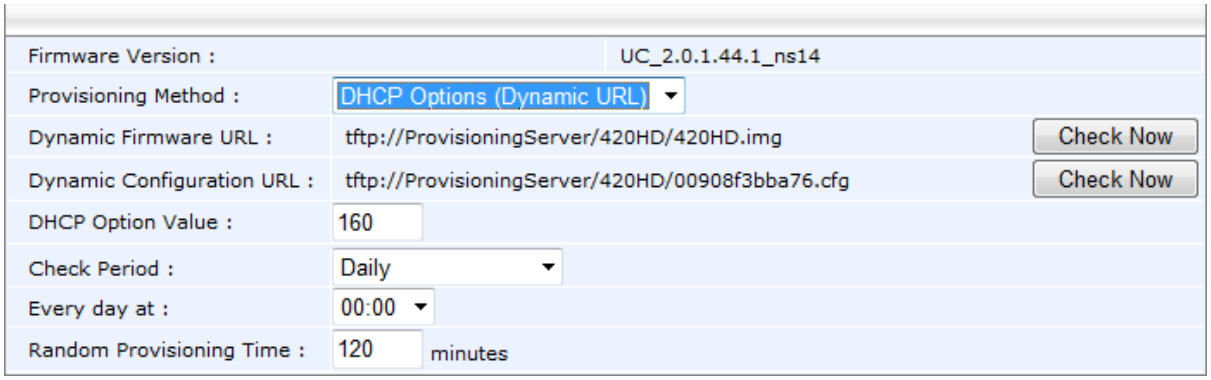
Dynamic Host Configuration Protocol (DHCP) can be used for automatic provisioning. Configure automatic provisioning with DHCP using the Web interface or the configuration file.

➢ To configure automatic provisioning with DHCP:

- **Using the Web interface:**

1. Access the Automatic Update page (**Management** tab > **Automatic Update** menu > **Automatic Provisioning**).

Figure 4-1: Web Interface - Configuring Automatic Provisioning by DHCP



2. Configure the parameters using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 4-1: Configuring Automatic Provisioning Performed by DHCP

Parameter	Description
Note: To add a value to these parameters, enter **`provisioning/`** followed by the parameter name equals the value (e.g. `provisioning/method=dynamic`).	
Provisioning Method **[provisioning/method]**	Defines the provisioning method: ▪ **[Disable]** Disable - Automatic update is disabled. The phone attempts to upgrade its firmware and configuration ▪ **[Dynamic]** DHCP Options (Dynamic URL) (default) - Using DHCP options 160 or 66/67 for provisioning ▪ **[Static]** Static URL - Using Static URL for provisioning
DHCP Option Value **[provisioning/url_option_value]**	Determines the DHCP option number to be used for receiving the URL for provisioning. The default value is 160. The phone supports DHCP Option 160 for complete URL and Options 66/67 for TFTP usage. Option 160 has the highest priority and if absent, Options 66/67 are used. The following syntax is available for DHCP option 160: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<firmware file name> ▪ <protocol>://<server IP address or host name>/<firmware file name>;<configuration file name> ▪ <protocol>://<server IP address or host name>/;<configuration file name> Where <protocol> can be one of the following: "ftp", "tftp", "http" or "https". For example: ▪ **ftp://192.168.2.1** – retrieved firmware file is *420HD.img* and the configuration file name is <MAC address>.cfg. For example, 001122334455.cfg ▪ **tftp://192.168.2.1/different_firmware_name.img** - retrieved firmware file is Different_Firmware_Name.img and the configuration file name is <MAC address>.cfg. For example, 001122334455.cfg ▪ **http://192.168.2.1/different_firmware_name.img ; 420HD_<MAC>_conf.cfg** - retrieved firmware file is different_firmware_name.img and the configuration file name is 420HD_<MAC address>_conf.cfg. For example, **420HD_001122334455_conf.cfg** • **https://192.168.2.1/420HD_<MAC>_conf.cfg** - In this case the retrieved firmware file is 420HD.img and the configuration file name is 420HD_<MAC

Parameter	Description
	Address>_conf.cfg. For example, 420HD_001122334455_conf.cfg The following syntax is available for DHCP Options 66/67: ▪ Option 66 must be a valid IP address or host name of a TFTP server only. ▪ Option 67 must be the firmware name. If Option 67 is absent, the phone requests for the 420HD.img image file. For example: ▪ **Option 66:** 192.168.2.1 or myTFTPServer ▪ **Option 67:** 420HD_1.2.2_build_5.img **Notes:** ▪ This parameter is applicable only when **method** is configured to "Dynamic". It is recommended to leave the parameter at its default value to avoid conflict with other DHCP options settings.
Random Provisioning Time **[provisioning/random_provisioning_time]**	Defines the maximum random number to start the provisioning process. This is used for periodic checking of firmware and configuration files to avoid multiple devices from starting the upgrade process at the same time. When the device is meant to start the upgrade, the device randomly selects a number between 1 and the value set for **random_provisioning_time** and performs the check only after the random time. The valid range is 0-65535. The default value is 120.
Check Period **[provisioning/period/type]**	Defines the period type for automatic provisioning: ▪ **[hourly]** Hourly **-** Sets an interval in hours. ▪ **[daily]** Daily (default) - Sets an hour in the day. ▪ **[weekly]** Weekly - Sets a day in the week and an hour in the day. ▪ **[powerup]** On Power-up Only - The phone tries to upgrade only after power-up.
Every (Check Period = Hourly) **[provisioning/period/hourly/hours_interval]**	The interval in hours for automatically checking for new firmware and configuration files. The valid range is 1 to 168. The default is 24. **Note:** This parameter is applicable only when **`type`** is configured to "hourly".
Every day at **[provisioning/period/daily/time]**	The hour in the day for automatically checking for new firmware and configuration files. The format of this value is **hh:mm**, where **hh** is hour and **mm** is minutes. For example, **`00:30.`** The default time is **00:00**. **Note:** This parameter is applicable only when **type** is configured to "daily".
Every (Check Period = Day) **[provisioning/period/weekly/day]**	The day in the week for automatically checking for new firmware and configuration files.

Parameter	Description
	▪ **[Sunday]** Sunday (default) ▪ **[Monday]** Monday ▪ **[Tuesday]** Tuesday ▪ **[Wednesday]** Wednesday ▪ **[Thursday]** Thursday ▪ **[Friday]** Friday ▪ **[Saturday]** Saturday **Note:** This parameter is applicable only when **type** is configured to "weekly".
Every (Check Period = Weekly) **[provisioning/period/weekly/time]**	The hour in the day for automatically checking for new firmware and configuration files. The format of this value is: **hh:mm**, where **hh** is hour and **mm** is minutes. For example: `00:30` The default time is **00:00**. **Note:** This parameter is applicable only when **type** is configured to "weekly".
provisioning/XXX_uri	The URI that allows the administrator to retrieve relevant information from separate files. The URI's can include specific protocol and path of the provisioning server or alternatively, the files can be retrieved from the server whose information was provided during provisioning (e.g. DHCP options 66/67 or 160). For example, in the below setting, *YYY.ext* is retrieved from the server whose information was provided during provisioning (e.g. DHCP options 66/67 or 160: **provisioning/XXX_uri=YYY.ext** Alternatively, the full URL and protocol can be set as follows: **provisioning/XXX_uri=<protocol>://<full path>/YYY.ext** For example: ▪ provisioning/XXX_uri=tftp://10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=ftp:// 10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=http:// 10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=https:// 10.10.10.10/YYY.ext

4.1.1 Configuring DHCP Option 160

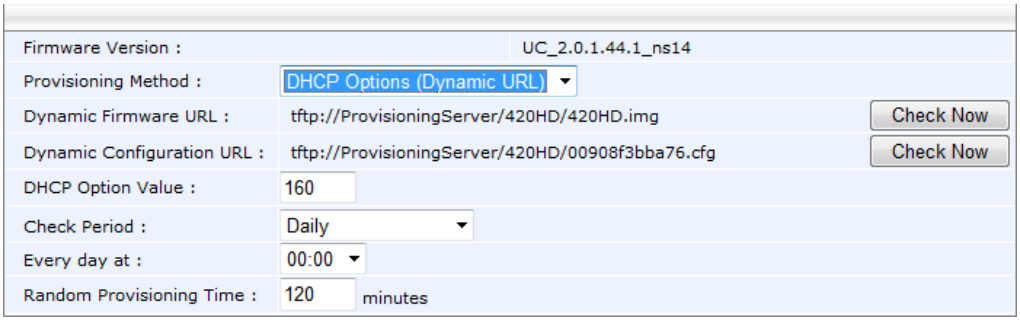
DHCP Option Value 160 can be configured using the Web as shown below.

➢ **To configure DHCP Option 160:**

■ **Using the Web interface:**

1. Access the Automatic Provisioning page (**Management** tab > **Automatic Update** menu > **Automatic Provisioning**).

Figure 4-2: Web Interface - Automatic Provisioning - DHCP Option 160



2. From the 'Provisioning Method' drop-down list, select **DHCP Option (Dynamic URL)**.
3. In the 'DHCP Option Value' field, enter **160**.
4. Configure the remaining parameters and click **Submit**.
5. After reboot, confirm that the firmware and configuration files have been updated.

4.1.2 Configuring DHCP Options 66 and 67

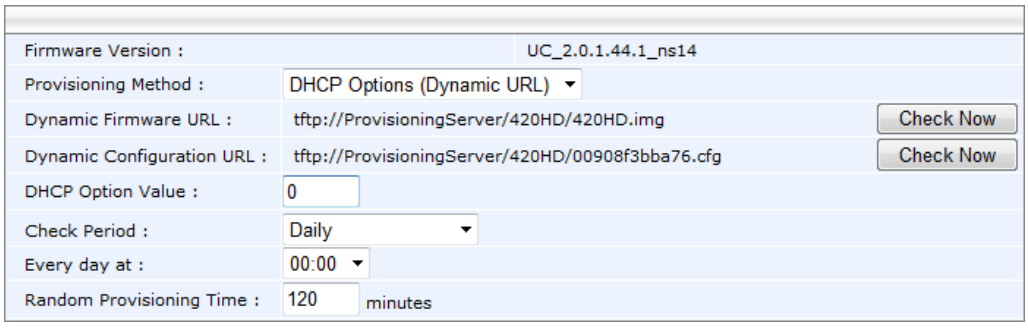
DHCP Options 66 and 67 can be configured using the Web as shown below.

➢ **To configure DHCP Options 66 and 67:**

■ **Using the Web interface:**

1. Access the Automatic provisioning page (**Management** tab > **Automatic Update** menu > **Automatic Provisioning**).

Figure 4-3: Web Interface – Automatic Provisioning - DHCP Options 66/67



2. From 'Provisioning Method', select **DHCP Option (Dynamic URL)**.
3. In the 'DHCP Option Value' field, enter **0**.
4. Configure the remaining parameters and click **Submit**.
5. After the reboot, confirm that the firmware and configuration files were updated.

4.2 Configuring Automatic Provisioning via Static DNS Record

The Static DNS (Generic Domain Name) Record method is used for automatic provisioning when you are unable to manage your DHCP server. If the provisioning server does not support using SIP SUBSCRIBE and NOTIFY messages mechanism as described above and no response for the SIP SUBSCRIBE message has been received, the phone tries to retrieve firmware and configuration files using the following URL:
tftp://ProvisioningServer/<*Phone Model Name*>/

For example:

- The phone tries to obtain the following firmware file:
  tftp://ProvisioningServer/420HD/420HD.img
- The phone tries to obtain the following configuration file:
  tftp://ProvisioningServer/420HD/<MAC address>.cfg

  (e.g. tftp://ProvisioningServer/420HD/001122334455.cfg)

It is the Administrator's responsibility to configure a DNS entry called **ProvisioningServer** on the DNS server and set it to the TFTP server IP address.

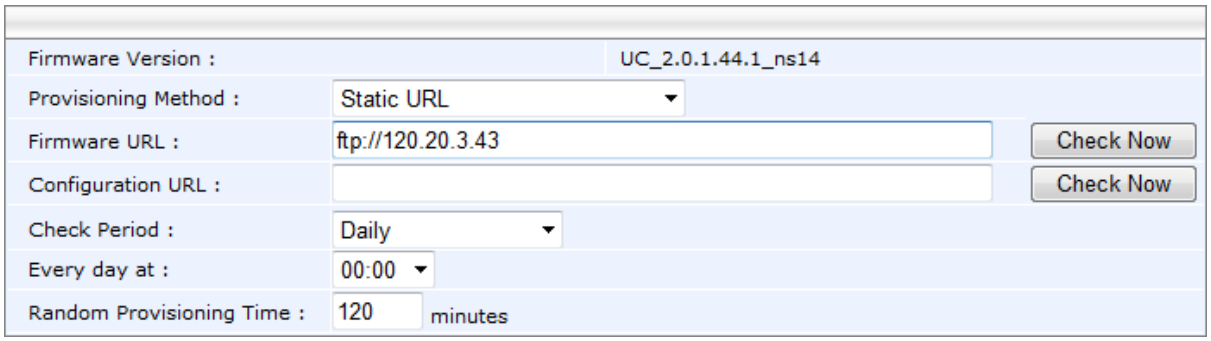
Note: If Generic Domain Name is used, the automatic provisioning mechanism periodically tries to retrieve new firmware/configuration from Provisioning Server domain name.

➢ To configure Static DNS Record:

- **Using the Web interface:**

  1. Access the Automatic Update page (**Management** tab > **Automatic Update** menu > **Automatic Provisioning**).

Figure 4-4: Web Interface - Static DNS Record



  2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 4-2: Static DNS Record Parameters

Parameter	Description
Firmware URL **[provisioning/firmware/url]**	The static URL for checking the firmware file. The URL must be entered using one of the following syntax options: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<firmware file name> Where <protocol> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example: ▪ tftp://192.168.2.1 – retrieved firmware file is 420HD.img ▪ ftp://192.168.2.1/Different_Firmware_Name.img - retrieved firmware file is Different_Firmware_Name.img **Note:** This parameter is applicable only when **method** is configured to "Static".
Configuration URL **[provisioning/configuration/url]**	The static URL for checking the configuration file. The URL must be entered using one of the following syntax options: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<configuration file name> Where <protocol> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example: ▪ http://192.168.2.1 - configuration file name is <MAC Address>.cfg, for example, 001122334455.cfg ▪ https://192.168.2.1/420HD_<MAC>_conf.cfg - retrieved configuration file name is 420HD_<MAC Address>_conf.cfg, for example, 420HD_001122334455_conf.cfg **Note:** This parameter is applicable only when **method** is configured to "Static".

4.3 Configuring Automatic Provisioning via SIP SUBSCRIBE and NOTIFY

If the provisioning information (e.g. Option fields 66/67/160) is not provided by the DHCP server, the phone sends a SIP SUBSCRIBE message to the multicast address **224.0.1.75:5060** as shown below.

Note: If the provisioning server supports using SIP SUBSCRIBE and NOTIFY messages and the device receives the provisioning URL in the NOTIFY message, the automatic provisioning mechanism then periodically tries to retrieve a new firmware/configuration according to the information provided.

```
SUBSCRIBE sip:224.0.1.75:5060 SIP/2.0
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
To: <sip:224.0.1.75:5060>
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
CSeq: 1 SUBSCRIBE
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Expires: 0
Event: ua-profile;profile-
type="application";model="420HD";version="1.2.1_build_6"
Max-Forwards: 70
Supported: replaces,100rel
Accept:  application/url
Contact: <sip:00000001@10.13.2.37:5060>
User-Agent: AUDC-IPPhone/1.2.1_build_6
Content-Length: 0
```

The provisioning server or any other entity replies with a 200 OK message to the SUBSCRIBE message (see below) and sends a NOTIFY SIP message with the provisioning URL in the message body as shown below.

If no response is received by the provisioning server, the phone resends SUBSCRIBE messages for five seconds.

With the above method, the phone uses its built-in auto-provisioning mechanism while the provisioning information is retrieved through the NOTIFY message.

The following code describes **SIP 200 OK Response on the SUBSCRIBE Message**:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Contact: <sip:10.13.2.37:5060>
To: <sip:224.0.1.75:5060>
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
CSeq: 1 SUBSCRIBE
Expires: 0
Content-Length: 0
```

The following code describes **SIP NOTIFY Message with Provisioning Information**.

```
NOTIFY sip:10.13.2.37:5060 SIP/2.0
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Max-Forwards: 20
Contact: <sip:10.13.4.121:5060>
To: <sip:224.0.1.75:5060>
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
CSeq: 1 NOTIFY
Content-Type: application/url
Subscription-State: terminated;reason=timeout
Event: ua-profile;profile-
type="application";model="420HD";version="1.2.0"
Content-Length: 18
tftp://10.13.4.121
```

The following code describes **SIP SUBSCRIBE Message to Obtain Provisioning Information.**

```
SUBSCRIBE sip:224.0.1.75:5060 SIP/2.0
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
To: <sip:224.0.1.75:5060>
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
CSeq: 1 SUBSCRIBE
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Expires: 0
Event: ua-profile;profile-
type="application";model="420HD";version="1.2.1_build_6"
Max-Forwards: 70
Supported: replaces,100rel
Accept:  application/url
Contact: <sip:00000001@10.13.2.37:5060>
User-Agent: AUDC-IPPhone/1.2.1_build_6
Content-Length: 0
```

The provisioning server or any other entity replies with a 200 OK message to the SUBSCRIBE message (see below) and sends a NOTIFY SIP message with the provisioning URL in the message body as shown below.

If no response is received by the provisioning server, the phone resends SUBSCRIBE messages for 5 seconds.

With the above method, the phone uses its built-in auto-provisioning mechanism while the provisioning information is retrieved through the NOTIFY message.

The following code describes **SIP 200 OK Response on the SUBSCRIBE Message.**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Contact: <sip:10.13.2.37:5060>
To: <sip:224.0.1.75:5060>
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
```

```
CSeq: 1 SUBSCRIBE
Expires: 0
Content-Length: 0
```

The following code describes **SIP NOTIFY Message with Provisioning Information.**

```
NOTIFY sip:10.13.2.37:5060 SIP/2.0
Via: SIP/2.0/UDP 10.13.2.37:5060;rport;branch=z9hG4bK-386d4398-
6ad00ca2-7ca3606e
Max-Forwards: 20
Contact: <sip:10.13.4.121:5060>
To: <sip:224.0.1.75:5060>
From: <sip:00000001@10.13.2.37:5060>;tag=87a5a8-25020d0a-13c4-
50029-386d4398-66dc40c-386d4398
Call-ID: 8884c8-25020d0a-13c4-50029-386d4398-3e2bcb8e-386d4398
CSeq: 1 NOTIFY
Content-Type: application/url
Subscription-State: terminated;reason=timeout
Event: ua-profile;profile-
type="application";model="420HD";version="1.2.0"
Content-Length: 18
tftp://10.13.4.121
```

5 Viewing Status and Monitoring

5.1 Viewing Network Status

This section shows how to view network status from the Web interface.

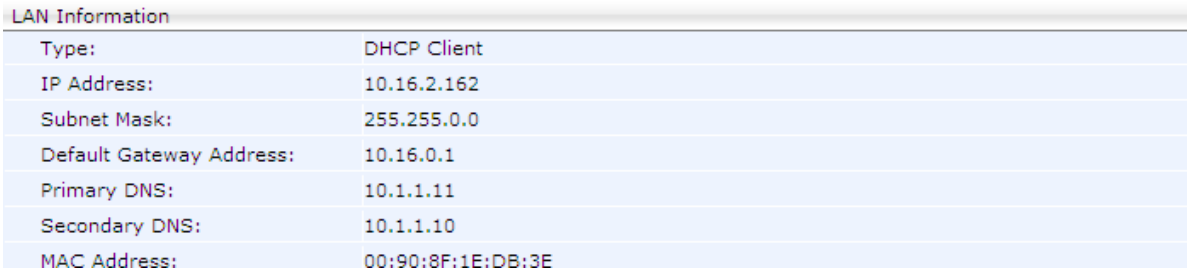
5.1.1 Viewing LAN Status

This section shows how to view LAN status information.

➢ **To view LAN status information:**

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **Network Status**).

Figure 5-1: Web Interface - LAN Information



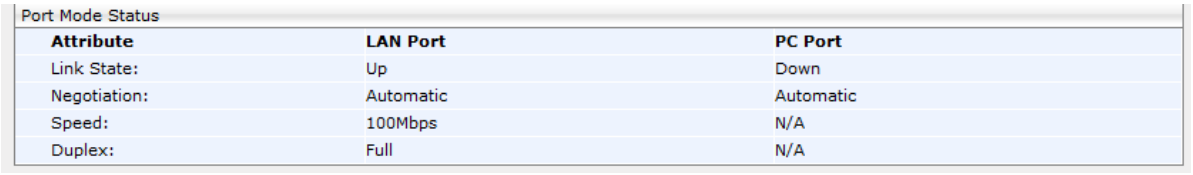
5.1.2 Viewing Port Status

This section shows how to view the Port Mode status.

➢ **To view port mode status:**

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **Network Status**).

Figure 5-2: Web Interface - Port Mode Status



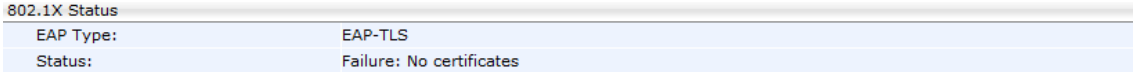
5.1.3 Viewing 802.1X Status

This section shows how to view 802.1X status.

➢ **To view 802.1X status:**

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **Network Status**).

Figure 5-3: Web Interface - 802.1X Status



5.2 Viewing VoIP Status

This section shows how to view VoIP status using the Web interface.

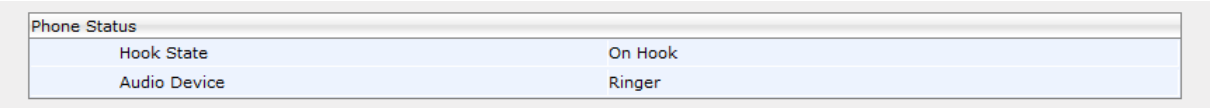
5.2.1 Viewing Phone Status

This section shows how to view the phone status.

➢ To view the phone status:

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **VoIP Status**).

Figure 5-4: Web Interface - Phone Status



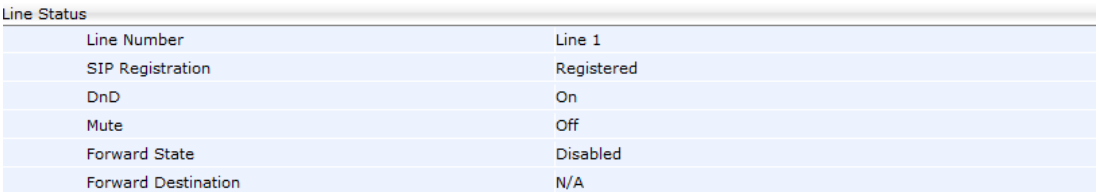
5.2.2 Viewing Line Status

This section describes how to view the line status.

➢ To view the line status:

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **VoIP Status**).

Figure 5-5: Web Interface - Line Status



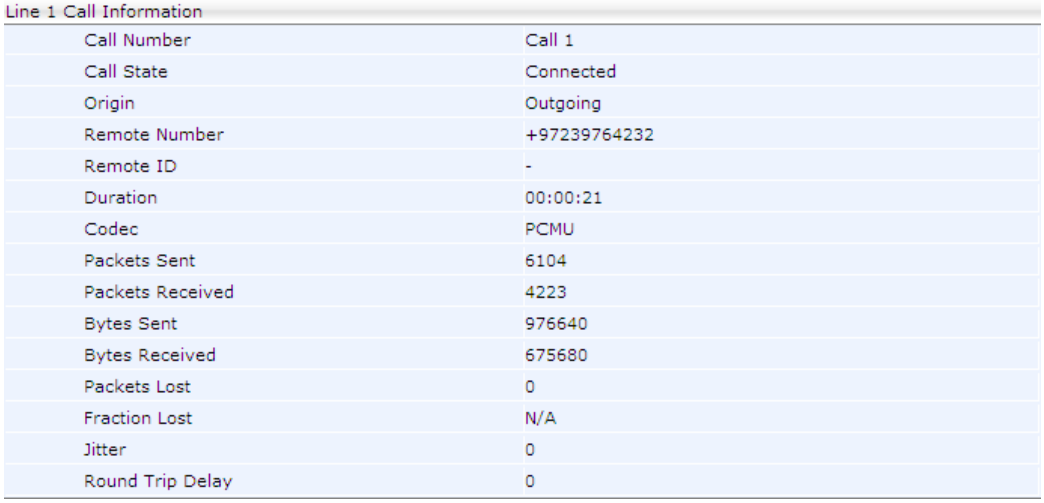
5.2.3 Viewing Call Information

The Web interface displays call information of a currently established call.

➢ **To view call information:**

- Access the Network Status page (**Status & Diagnostics** tab > **System Status** menu > **VoIP Status**).

Figure 5-6: Web Interface - Call Information



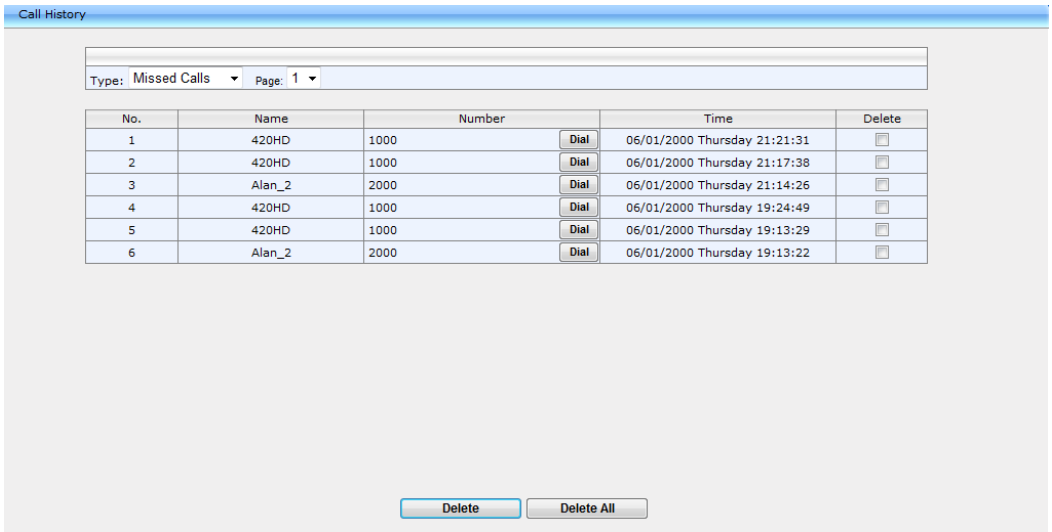
5.3 Viewing Calls History

The Web interface displays received and missed calls, dialed numbers and call duration.

➢ **To view calls history:**

1. Access the Call History page (**Status & Diagnostics** tab > **History** > **Call History**).

Figure 5-7: Web Interface - Call History



2. From the 'Type' drop-down list, select the type of call history (i.e., missed calls, received calls, and dialed numbers) that you want to view; the table lists the call history according to the chosen call history type.
3. To delete an entry, select the entry's 'Delete' option and click the **Delete** button.

5.4 Viewing Phone Model / Firmware Version

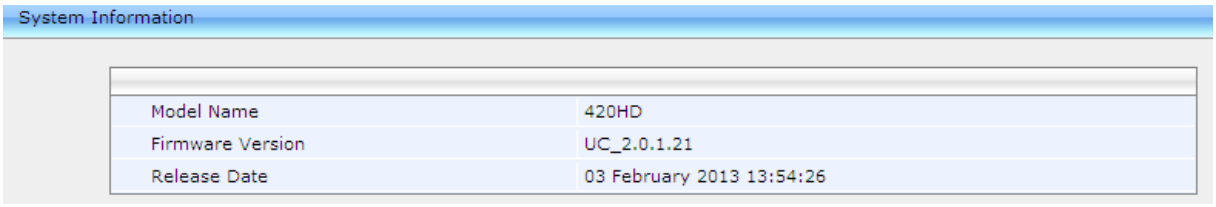
This section shows how to view the phone model and the phone's firmware version from the Web interface or from the phone's LCD.

5.4.1 Viewing from the Web Interface

➢ **To view the phone's model and firmware version from the Web interface:**

- Access the phone's Web interface; the System Information page opens by default:

Figure 5-8: Web Interface - System Information



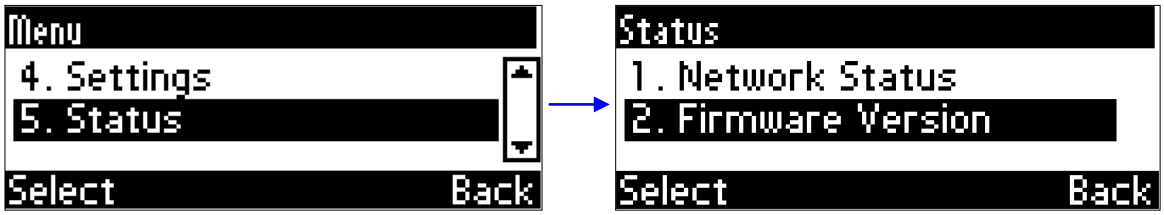
To access the page from another page in the Web interface:
Status & Diagnostics tab > **System Information** menu > **Versions**

5.4.2 Viewing from the Phone's LCD

This section shows how to view phone model and firmware version from the phone's LCD.

➢ **To view the phone's model and firmware version from the LCD:**

- Access the Firmware Version screen (MENU key > **Status** menu > **Firmware Version**).



6 Troubleshooting

This section provides various troubleshooting procedures.

6.1 Unable to Sign in to Lync using Username/Password

Problem	Unable to sign in to Lync using the username/password sign-in method.
LCD Message	"Invalid address, username or password"
Corrective Actions	
1 Ensure that you have correctly entered the sign-in address, username, and password. 2 Ensure that you have the correct username/password; it may have changed in the Enterprise's Active Directory. 3 Ensure that you are in the correct sign-in method (press the **Sign-in** softkey, scroll down to **Switch sign-in method**, and then press the OK key or **Select** softkey).	

6.2 Unable to Authenticate User using PIN

Problem	Unable to authenticate user when signing in to Lync using PIN code.
LCD Message	"The phone number or extension is not valid"
Corrective Actions	
1 Ensure that you have entered the phone number / PIN code correctly. 2 Ensure that you have the correct PIN code; it may have changed in the Enterprise's Active Directory. 3 Ensure that you are in the PIN code sign-in method (press the **Sign-in** softkey, scroll down to **Switch sign-in method**, and then press the OK key or **Select** softkey).	

6.3 IP Phone Fails Registration Process

Problem	The phone fails to register.
LCD Message	-
Corrective Actions	
Ensure the following: 1 DHCP Option 43 has been configured. 2 Access is possible from the following Web site: https://YOUR_AUTHORITY_SERVER:443/CertProv/CertProvisioningService.svc 3 If the environment supports more than one CA Certificate, this must be included in the CA Certificate file and loaded to the IP Phone.	

6.4 How to Verify CA Certificate is Trusted / Authorized by IP Phone

Problem	How do I know if my CA Certificate is trusted and authorized by the IP Phone?
LCD Message	-
Corrective Actions	
Verify whether your public trusted certificate is listed in Microsoft Public Trusted Certificates (http://technet.microsoft.com/en-us/library/gg398270(v=ocs.14).aspx).	

6.5 Invalid Time Server

Problem	The time server is invalid.
LCD Message	-
Corrective Actions	
Ensure that NTP (DHCP Option 42) is configured in the DHCP server and is defined as NTP SRV records. If not, manually configure it.	

6.6 Invalid Time Offset

Problem	The time offset is incorrect.
LCD Message	-
Corrective Actions	
Ensure that the Time Offset (DHCP Option 2) is configured in the DHCP server. If not, manually configure Daylight Saving Time (DST) values in the 420HD IP Phone.	

6.7 General Corrective Actions

6.7.1 Restoring Phone Defaults

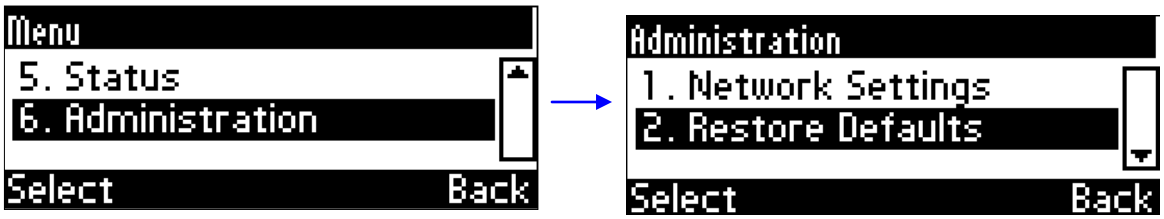
The phone's default settings can be restored from the phone's LCD or from the Web interface.

6.7.1.1 Restoring Factory Defaults from the Phone LCD

This section shows how to restore factory defaults from the phone's LCD.

➢ **To restore the phone's default settings from the LCD:**

1. Access the **Restore Defaults** menu option (**MENU** key > **Administration** > **Restore Defaults**):



2. Press the **Select** softkey:

3. Press the **Yes** softkey to confirm or **No** to cancel.

Note: With the 420HD IP Phone you can also restore the phone to default settings by using a hard-coded sequence of keys on the phone's keypad:

1. Unplug the power cable from the phone.
2. Simultaneously press the **OK** and the **MENU** key.
3. Plug the power cable into the phone and keep the keys pressed for five seconds.

6.7.1.2 Restoring Factory Defaults from the Web Interface

This section shows how to restore the phone's factory defaults from the Web interface.

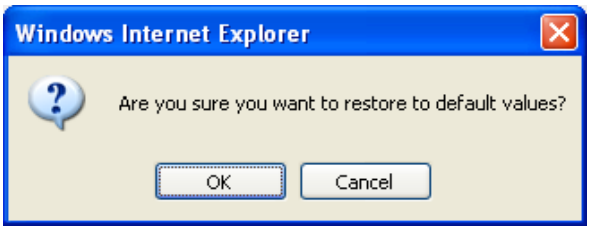
➢ **To restore the phone's factory defaults from the Web interface:**

1. Access the Restore Defaults page (**Management** tab > **Administration** > **Restore Defaults**).

Figure 6-1: Web Interface - Restore Defaults

2. Click the **Submit** button:

Figure 6-2: Confirm Restore to Factory Defaults



3. Click **OK**.

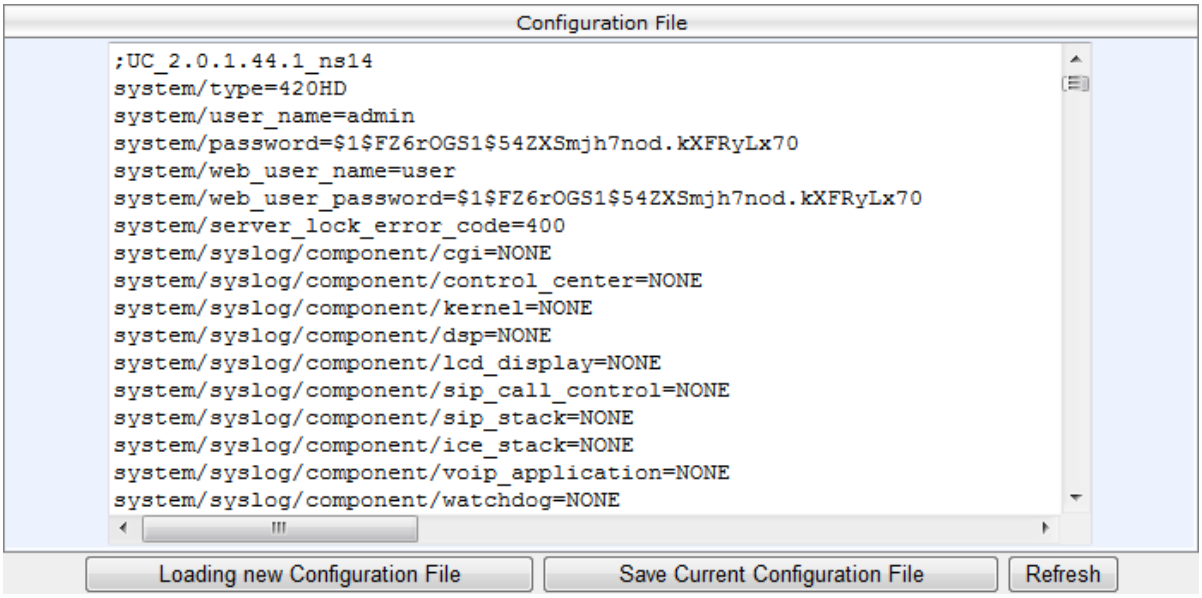
6.7.2 Loading the Configuration File Manually

This section shows how to load the cfg configuration file to the phone.

➢ **To load the cfg configuration file to the phone:**

1. In the Web interface, access the Configuration File page (**Management** tab > **Manual Update** menu > **Configuration File**):

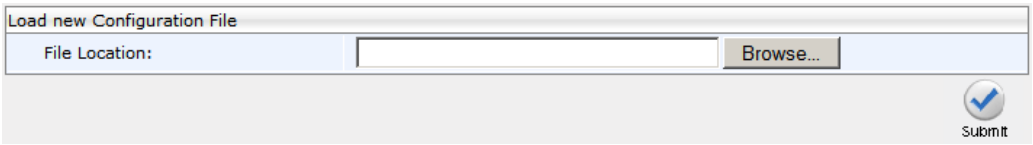
Figure 6-3: Web Interface - Configuration File



The configuration you created is displayed in the text pane.

2. Click the **Loading new Configuration File** button:

Figure 6-4: Web Interface - Load New Configuration File



3. Click **Browse** and select the cfg file you created; the phone verifies that it's related to the phone model. The cfg is then loaded to the phone. Once loaded, the phone reboots (indicated on the LCD). The phone is now loaded with the cfg configuration file you created.

6.7.3 Recovering Firmware

If the phone is powered off for some reason during the firmware upgrade process, the phone becomes unusable.

➢ **To recover the phone firmware:**

1. Ensure that your DHCP server supports Options 66 (TFTP server address) and 67 (firmware file), and that these are configurable.
2. Before connecting the phone, verify that the TFTP server is running and the firmware file for recovery is located in the correct location.
3. Connect your phone to the IP network, and then connect the phone to the power outlet;
   a. The phone sends a TFTP request to the IP address indicated in the DHCP Option 66 field to retrieve the firmware file indicated in the DHCP Option 67 field.
   b. The phone, in the DHCP Discover message sends its model name in the DHCP Option 77 field. The DHCP server, according to the phone model, sets the appropriate firmware file name in the DHCP Option 67 field sent to the phone (e.g., 420HD_1.4.1_build_32.img).
   c. The phone then upgrades to the recovery firmware.
   d. After the firmware upgrade process completes, the phone boots up successfully.

6.7.4 Restarting the Phone

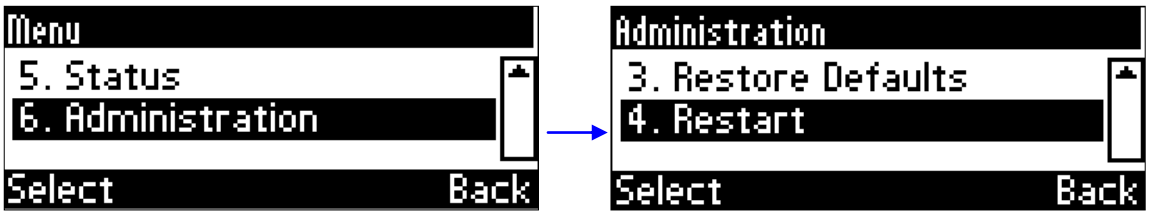
The IP Phone can be restarted from phone's LCD or the Web interface.

6.7.4.1 Restarting the Phone from the LCD

This section shows how to restart the phone from its LCD.

➢ **To restart the phone from its LCD:**

1. Select the **Restart** submenu (**MENU** key > **Administration** > **Restart**):



A warning message appears requesting you to confirm:

2. Press the **Yes** softkey to confirm phone restart or **No** to cancel.

6.7.4.2 Restarting the Phone from the Web Interface

This section shows how to restart the phone from the Web interface:

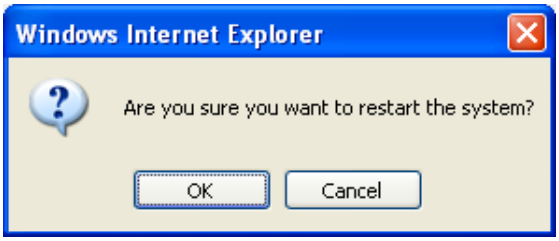
➢ **To restart the phone from the Web interface:**

1. Access the Restart System page (**Management** tab > **Administration** menu > **Restart System**).

Figure 6-5: Web Interface - Restart System

2. Click the **Restart** button:

Figure 6-6: Confirmation Box



3. Click **OK**.

6.7.5 Analyzing and Debugging Traffic using Syslog

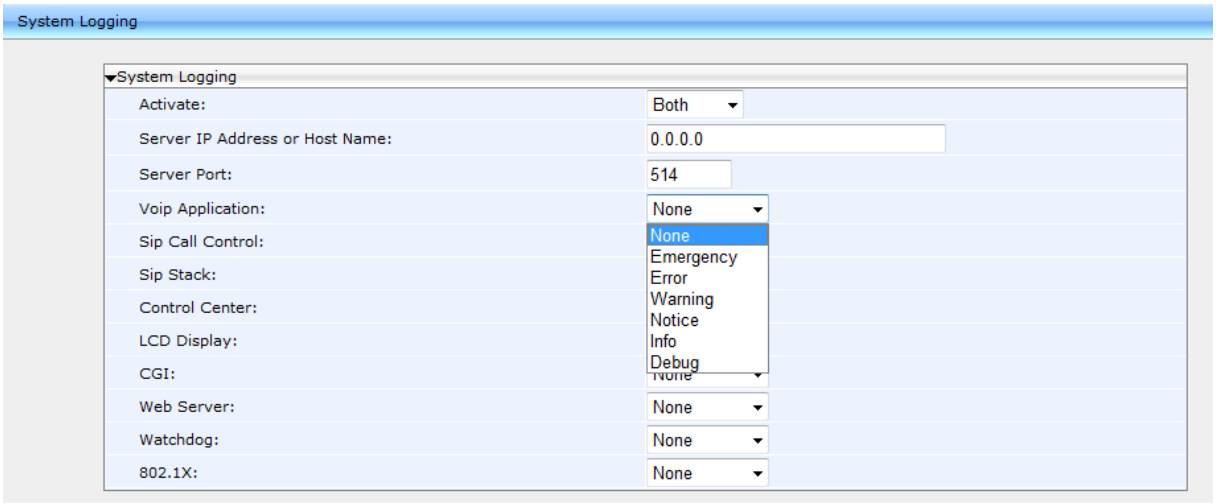
System logging (Syslog) is used for traffic analysis and debugging. The feature can be configured from the Web interface or from the configuration file.

➢ To configure system logging:

- **Using the Web interface:**

1. Access the System Logging page (**Status & Diagnostics** tab > **Diagnostics** menu > **Logging**).

Figure 6-7: Web Interface - System Logging



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 6-1: Syslog Parameters

Parameter	Description
Note: To add a value to these parameters, enter **system/** followed by the parameter name, equal sign and then the value (e.g. system/syslog/mode=UDP).	
Activate **[system/watchdog/enabled]**	Enables the system watchdog. Possible values are: • **Disable** = No Syslog. • **Network** = Syslog is sent to the Syslog server. (Recommended). • **Console** = Syslog is sent to the phone console. (You need to connect a serial cable to view the logs. This causes delays in phone operation). • **Both** = Syslog sends to the Syslog server AND the console. **Note:** It is recommended to leave this parameter at its default value.

Parameter	Description
Server IP Address or Host Name **[system/syslog/server_address]**	The IP address (in dotted-decimal notation) of the computer you are using to run the Syslog server (e.g. Wireshark). The Syslog server is an application designed to collect the logs and error messages generated by the phone. The default IP address is 0.0.0.0. **Note:** This parameter is applicable when Activate is set to "Network" or "Both".
Server Port **[system/syslog/server_port]**	Defines the UDP port of the Syslog server. The valid range is 0 to 65,535. The default port is 514. **Note:** This parameter is applicable when Activate is set to "Network" or "Both".
Note: The following Severity level options are applicable for the fields below: • None • Emergency • Error • Waiting • Notice • Info • Debug	
VoIP Application	Defines multi-layer VoIP application.
SIP Call Control	Defines MTR layer Radvision.
SIP Stack	Defines SIP Stack Radvision.
Control Center	Responsible for Networking and running other processes.
LCD Display	Defines LCD Display.
CGI	Defines services for the Web server.
Web Server	Defines the phone Web server.
Watchdog	Responsible for keeping other processes running.
802.1X	Defines the security protocol.

6.7.6 Enabling Recording to Debug Voice

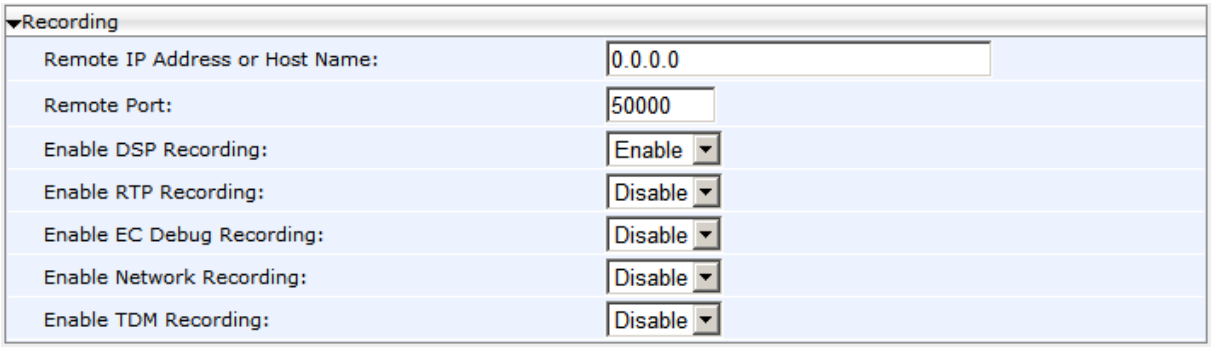
Recording capability enables you to debug the voice activity of the phone. Enable the feature from the Web interface or using the configuration file.

➢ To define media debug information:

- **Using the Web interface:**

1. Access the Recording page (**Status & Diagnostics** tab > **Diagnostics** menu > **Recording**).

Figure 6-8: Web Interface - Recording



2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 6-2: Packet Recording Parameters

Parameter	Description
Enable DSP Recording **[voip/packet_recording/enabled]**	Activates the packet recording mechanism. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Remote IP Address or Host Name **[voip/packet_recording/remote_ip]**	The IP address (in dotted-decimal notation) of the remote computer to which the recorded packets are sent. The recorded packets should be captured by a network sniffer (such as Wireshark). The default value is 0.0.0.0.
Remote Port **[voip/packet_recording/remote_port]**	Defines the UDP port of the remote computer to which the recorded packets are sent. The valid range is 1024 to 65535. The default value is 50000.
Enable RTP Recording **[voip/packet_recording/rtp_recording/enabled]**	Activates the DSP RTP recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable

Parameter	Description
Enable EC Debug Recording **[voip/packet_recording/ec_debug_recording/enabled]**	Activates the Echo Canceller Debug recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Enable Network Recording **[voip/packet_recording/network_recording/enabled]**	Activates the DSP network (TDM Out) recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Enable TDM Recording **[voip/packet_recording/tdm_recording/enabled]**	Activates the DSP TDM (TDM In) recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable

6.7.7 Activating Port Mirroring

Port mirroring facilitates network traffic monitoring enabling administrators to analyze and debug data or diagnose errors on a network, manage network performance and be alerted to problems.

Inbound or outbound traffic (or both) can be mirrored on single or multiple interfaces.

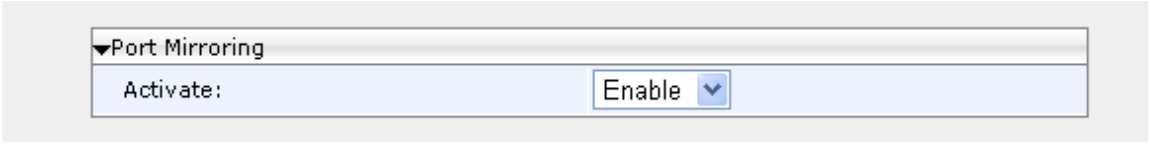
Port mirroring can be activated using the Web interface or the configuration file.

➢ To configure port mirroring:

- **Using the Web interface:**

  1. Access the Recording page (**Configuration** tab > **Network Connections** menu > **Network Settings**).

Figure 6-9: Web Interface - Port Mirroring



  2. Configure using the table below as reference and click **Submit**.

- **Using the configuration file:**

Table 6-3: Port Mirroring Parameters

Parameter	Description
Activate **[network/pc_port_mirroring/enabled]**	Enables port mirroring. ▪ **[0]** Disable (default) **-** The LAN/PC Network interfaces operate in SWITCH mode. ▪ **[1]** Enable **-** The LAN/PC Network interfaces operate in HUB mode. The network traffic on the LAN port is reflected in the PC port.

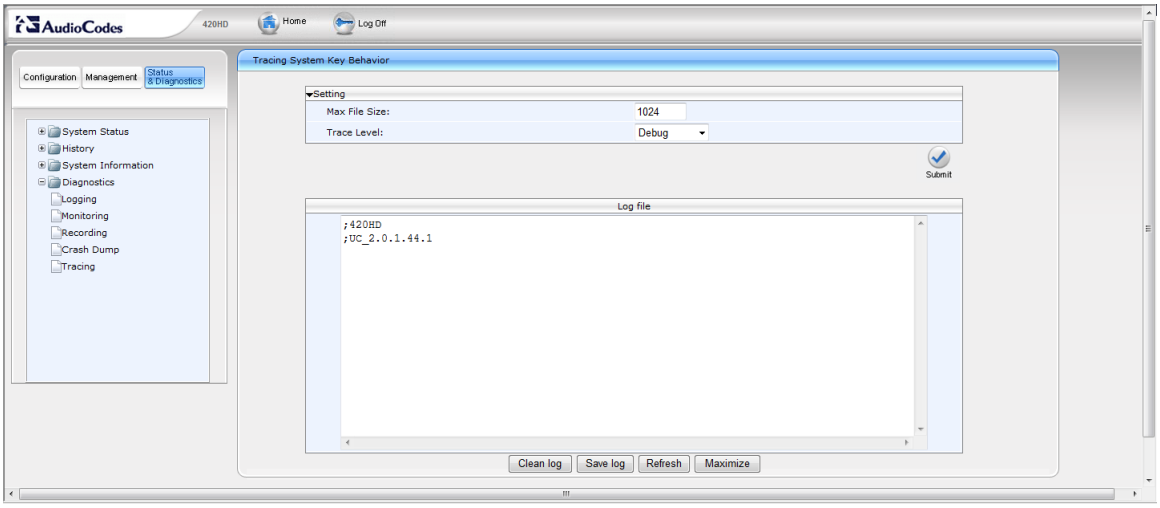
6.7.8 Enabling Tracing

It is recommended to set up the phone to store trace messages for effective troubleshooting and diagnosis.

➢ To enable tracing:

1. Log in to the Web management tool of the 420HD IP Phone, using HTTPS (e.g., https://172.17.0.103) through your Web browser. The phone's IP address can be obtained from the DHCP server, or from the 420HD LCD screen (press MENU button > press 5 for **Status** menu > press OK button for **Network Status** menu > press OK button > press the Navigation key to scroll down until the IP address is displayed).
2. Open the Tracing page (**Status & Diagnostics** tab > **Diagnostics** menu > **Tracing**).
3. Set the 'Max File Size' field to **1024**.
4. Set the 'Trace Level' field to **Debug** to activate tracing to debug level.
5. Click **Submit**.
6. Click **Clean log**, and then power up the 420HD IP Phone.
7. After the 420HD IP Phone completes the power-up process, click **Save log**.
8. Send the saved logged trace to AudioCodes Customer Technical Support for troubleshooting.

Figure 6-10: Configuring Tracing



A Product Specifications

The table below details the 420HD IP Phones specifications.

Table A-1: 420HD IP Phone Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling ▪ SIP over TLS (SIPS) ▪ 802.1p/Q for Traffic Priority and QoS ▪ VLAN Discovery Mechanism (CDP, LLDP) ▪ ToS (Type of Service) field, indicating desired QoS DHCP Client ▪ NTP Client
Media Processing	▪ Voice Coders: G.711, G.729A/B, G.722 ▪ Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length ▪ Adaptive Jitter Buffer 300 msec ▪ Voice Activity Detection ▪ Comfort Noise Generation ▪ Packet Lost Concealment ▪ RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC 3711) ▪ DTMF Relay (RFC 2833)
Telephony Features	▪ Call Hold / Un-Hold ▪ Call Transfer ▪ 3-Way Conferencing (with local mixing) ▪ Redial ▪ Caller ID Notification ▪ Call Waiting Indication ▪ Message Waiting Indication (including MWI LED) ▪ Local and Corporate Directories ▪ Automatic On-hook Dialing ▪ Automatic Answering (Alert-Info header and "talk" event) ▪ CWRR (Call Waiting Reminder Ring) ▪ Secondary Dial Tone ▪ Call Logs: Missed/Received Calls and Dialed Numbers ▪ Speed Dial (Including BLF support for Asterisk and BroadWorks™) ▪ URL Dialing ▪ Call Forward (Unconditional, Busy, No Answer) ▪ Remote Conferencing (RFC 4240)

Feature	Details
Configuration/ Management	▪ LCD Display User Interface Language Support (Various Languages) ▪ Web-based Management (HTTP/HTTPS) ▪ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and configuration file upgrade ▪ DHCP options (66, 67, and 160) for Auto-provisioning ▪ DHCP options (12, 60, and 77) for device information ▪ Private labeling mechanism ▪ Configuration file encryption (entire file and individual parameters)
Debugging Tools	▪ IPP Tracing ▪ Syslog mechanism ▪ DSP recording ▪ Port mirroring ▪ VoIP Status Web page
420HD Hardware	▪ **LCD screen:** Graphic LCD (132*64) mono ▪ **Connectors interfaces:** ✓ 2 x RJ-45 ports (10/100BaseT Ethernet) for WAN and LAN ✓ RJ-9 port (jack) for Headset ✓ RJ-9 port (jack) for Handset ▪ **Mounting:** ✓ Wall mounting ✓ Adjustable angle tilt for desktop mounting ▪ **Power:** ✓ DC jack adapter 12V ✓ Power supply AC 100 ~ 240V ✓ PoE: IEEE802.3af (optional) ▪ **Keys:** ✓ 4 x softkeys ✓ VOICE MAIL message hotkey ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE ✓ TRANSFER ✓ VOLUME control key ✓ HEADSET ✓ SPEAKER
Headset Compatibility	▪ GN.com 2100 – both single and dual ▪ Jabra BIZ 2400 – both single and dual ▪ GN.com 9350 ▪ Plantronics H261N ▪ Plantronics H251N

A.1 SIP Support (RFC, Headers)

The following is a list of supported SIP RFCs and methods that you can use to create for the phone.

Table A-2: Supported IETF RFCs

RFC Number	RFC Title
RFC 2327	SDP
RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
RFC 2782	A DNS RR for specifying the location of services
RFC 2833	Telephone event
RFC 3261	SIP
RFC 3262	Reliability of Provisional Responses in SIP
RFC 3263	Locating SIP Servers
RFC 3264	Offer/Answer Model
RFC 3265	(SIP)-Specific Event Notification
RFC 3310	Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)
RFC 3311 (Partially Supported)	UPDATE Method
RFC 3326 (Partially Supported)	Reason header
RFC 3389	RTP Payload for Comfort Noise
RFC 3515	Refer Method
RFC 3605	RTCP attribute in SDP
RFC 3611	RTP Control Protocol Extended Reports (RTCP XR)
RFC 3665	SIP Basic Call Flow Examples
RFC 3711	The Secure Real-time Transport Protocol (SRTP)
RFC 3725	Third Party Call Control
RFC 3842	MWI
RFC 3891	"Replaces" Header
RFC 3892 (Sections 2.1-2.3 and 3 are supported)	The SIP Referred-By Mechanism
RFC 3960 (Partially Supported)	Early Media and Ringing Tone Generation in SIP (partial compliance)
RFC 3966	The tel URI for Telephone Numbers
RFC 4028 (Partially Supported)	Session Timers in the Session Initiation Protocol
RFC 4240	Basic Network Media Services with SIP - NetAnn
draft-ietf-sip-privacy-04.txt (Partially Supported)	SIP Extensions for Network-Asserted Caller Identity using Remote-Party-ID header

RFC Number	RFC Title
draft-ietf-sipping-cc-transfer-05	Call Transfer
draft-ietf-sipping-realtimefax-01	SIP Support for Real-time Fax: Call Flow Examples
draft-choudhuri-sip-info-digit-00	SIP INFO method for DTMF digit transport and collection
draft-mahy-sipping-signaled-digits-01	Signaled Telephony Events in the Session Initiation Protocol

Note: The following SIP features are not supported:

- Preconditions (RFC 3312)
- SDP - Simple Capability Declaration (RFC 3407)
- S/MIME
- Outbound, Managing Client-Initiated Connections (RFC 5626)
- SNMP SIP MIB (RFC 4780)
- SIP Compression – RFC 5049 (SigComp)
- ICE (RFC 5245)
- Connected Identity (RFC 4474)

A.1.1 SIP Compliance Tables

The SIP device complies with RFC 3261 as shown in the following subsections.

A.1.1.1 SIP Methods

The device supports the following SIP methods:

Table A-3: Supported SIP Methods

Method	Supported	Comments
INVITE	Yes	
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	Send only
REFER	Yes	Inside and outside of a dialog
NOTIFY	Yes	
INFO	Yes	
OPTIONS	Yes	
PRACK	Yes	
PUBLISH	Yes	Send only
SUBSCRIBE	Yes	

A.1.1.2 SIP Headers

The device supports the following SIP headers:

Table A-4: Supported SIP Headers

Header Field	Supported
Accept	Yes
Alert-Info	Yes
Allow	Yes
Authorization	Yes
Call-ID	Yes
Call-Info	Yes
Contact	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Diversion	Yes
Encryption	No
Expires	Yes
Fax	Yes
From	Yes
History-Info	Yes
Join	Yes
Max-Forwards	Yes
MIN-SE	Yes
P-Asserted-Identity	Yes
P-Preferred-Identity	Yes
Proxy- Authenticate	Yes
Proxy- Authorization	Yes
Prack	Yes
Record- Route	Yes
Refer-To	Yes
Referred-By	Yes
Replaces	Yes
Remote-Party-ID	Yes
Retry-After	Yes
Route	Yes

Header Field	Supported
Session-Expires	Yes
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User- Agent	Yes
Via	Yes
Voicemail	Yes
Warning	Yes
WWW- Authenticate	Yes

Reader's Notes

AudioCodes Series of High Definition IP Phones

HD VoIP

Administrator's Manual

420HD IP Phones with Microsoft® Lync™ Server 2010

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